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PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant	:	Mohammed Javed Absar et al.
Application No.	:	09/622,736
Filed	:	October 27, 2000
For	:	FAST FREQUENCY TRANSFORMATION TECHNIQUE FOR TRANSFORM AUDIO CODERS

Examiner	:	Qi Han
Art Unit	:	2654
Docket No.	:	851663.413USPC
Date	:	July 10, 2006

Mail Stop Appeal Brief - Patents
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

SUPPLEMENTAL APPELLANTS' BRIEF

Commissioner for Patents:

This brief is in furtherance of the Notice of Appeal, filed in this case on February 16, 2005. This brief also is in response to a Notification of Non-Compliant Appeal Brief mailed on March 9, 2006. The Notification requested a discussion of each independent claim in the "Summary of Claimed Invention," a concise statement of each ground of rejection and a related proceedings appendix. In response, Applicants have revised the previously submitted brief as follows:

(1) Minor modifications have been made to the headings and language to more closely match the new headings and language set forth in 37 C.F.R. § 41.37(c)(1);

(2) Supporting references to page numbers and figures have been added where appropriate to the Summary of the Claimed Subject Matter;

(3) The previously submitted grounds for rejection for which review was sought have been restated;

(4) A copy of PCT Application No. PCT/SG98/00014 filed February 21, 1998 has been included;

(5) A copy of an International Preliminary Examination Report with an Article 34 Amendment filed with the U.S. National Phase Application has been included; and

(6) An Appendix indicating that there is no related-proceedings has been added.

The fees required under Section 1.17(c), and for a three-month extension of time, are enclosed with the accompanying fee transmittal. Applicants hereby request any additional fees necessary for acceptance of this Appeal Brief and any other fees which may become due be charged to Deposit Account No. 19-1090.

I. REAL PARTY IN INTEREST

The real party in interest is STMicroelectronics Asia Pacific PTE Limited, which is the assignee of the present invention. The assignment of record is to STMicroelectronics Asia Pacific PTE Limited, having an address at 28 ANG MO KIO Industrial Park 2, Singapore, Singapore 569508.

II. RELATED APPEALS AND INTERFERENCES

There are no other prior or pending appeals, interferences or judicial proceedings known to Appellants, Appellants' legal representative or assignee, which may be related to, directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

III. STATUS OF CLAIMS

Claims 1-39 are currently pending in this application. All pending active claims are attached hereto as Appendix A.

Claims 1-9 and 17-23 were rejected under 35 U.S.C. § 103(a) as obvious over U.S. Patent No. 5,479,562 ("Fielder"). Claims 10-13, 16, and 24-27 were rejected under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis, et al, Digital Signal Processing, principles, algorithms and applications (3d ed. 1996) ("Proakis"). Claims 14-15 and 28-39 were rejected under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis and U.S. Patent No. 6,304,847 ("Jhung").

The rejection of claims 1-39 is appealed.

IV. STATUS OF AMENDMENTS

No amendments were filed subsequent to the final rejection.

V. SUMMARY OF CLAIMED SUBJECT MATTER

This application is a conversion of PCT Application No. PCT/SG98/00014 filed February 21, 1998 into a U.S. National Application. The following summary discusses the subject matter of the appealed claims along with references to portions of the specification and drawings that provide support for the claims. The references are provided for exemplary purposes and are not intended to restrict the scope of the claims to the particular embodiments corresponding to the references provided. There are no means-plus-function or step-plus-function claims.

In general, the input to an audio coder comprises a stream of digitized samples of a time domain analog signal. For a multi-channel encoder, the stream consists of interleaved samples for each channel. The input stream is sectioned into blocks, each block containing N consecutive samples of each channel. See Figure 1 of the present application, reproduced below for convenience as Figure 1. Thus, within a block the N samples of a channel form a sequence {x[0], $x[1]$, $x[2]$. ..., $x[N-1]$}.

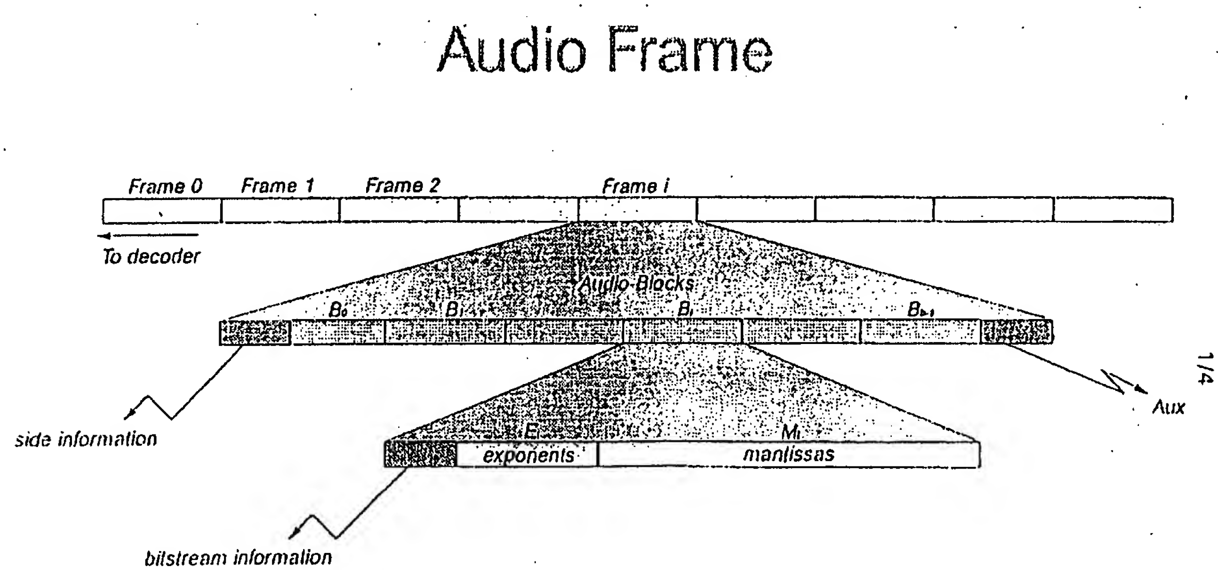

Fig. 1

The time domain samples are converted to the frequency domain using an analysis filter bank. The frequency domain coefficients, thus generated, form a coefficient set which can be identified as (X_0, X_1, X_2, ..., $X_{N/2-1}$). An example embodiment of a digital audio encoder is illustrated in Figure 2, which is reproduced below for convenience.

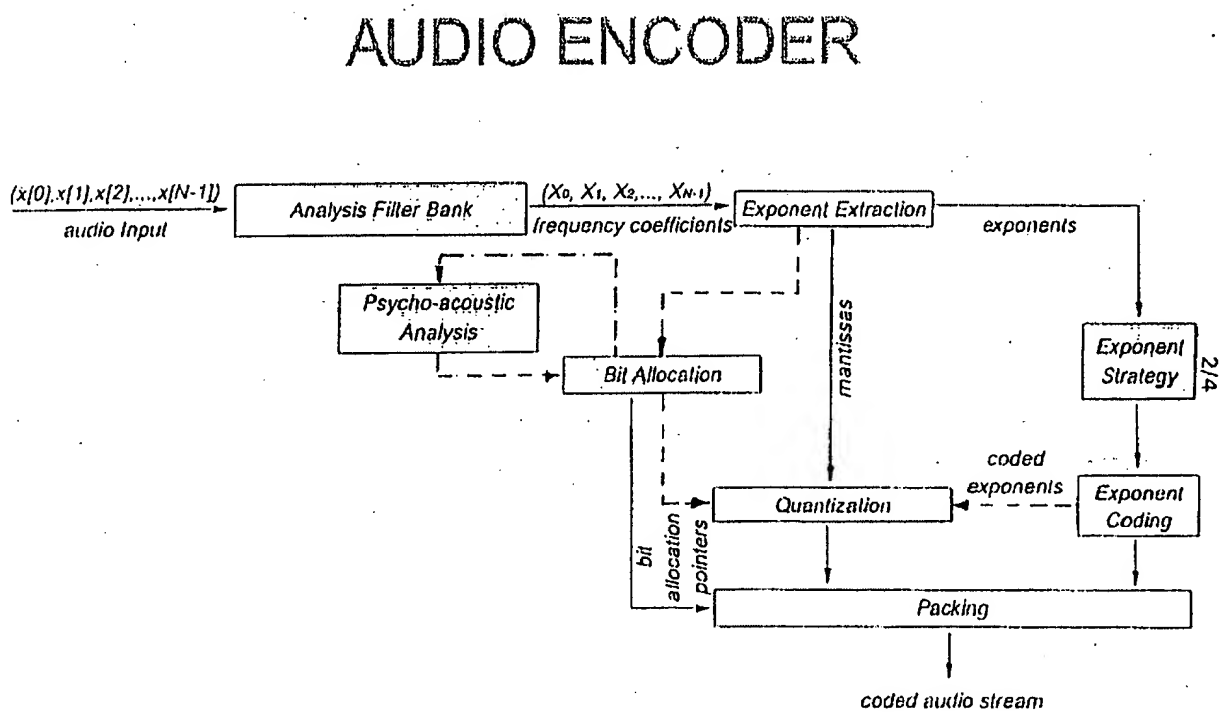

Fig. 2

X_0 is the lowest frequency component (DC) while $X_{N/2-1}$ is the highest frequency component of the signal. Only the first n/2 frequency components are considered because the signal is real. Audio compression essentially entails finding how much of the information in the set (X_0, X_1, X_2, ..., $X_{N/2-1}$) is necessary to reproduce the original analog signal at the decoder with minimal audible distortion.

The coefficient set is normally converted into floating point format, where each coefficient is represented by an exponent and mantissa. The exponent set is usually transmitted in its original form. The mantissa, however, is usually truncated to a fixed or variable number of decimal places. The value of the number of bits for coding a mantissa is usually obtained from a bit allocation algorithm, which for advanced psychoacoustic coders may be based on the masking properties of the human auditory system. A low number of bits results in a high compression ratio because less

space is required to transmit the coefficients. This may lead, however, to a high quantization error, leading to audible distortion. A good distribution of available bits to each mantissa forms the core of most advanced encoders.

One embodiment is directed to a method for encoding audio data in which coded Fast Modified Discrete Cosine Transform (FMDCT) coefficients are computed utilizing a Fast Fourier Transform (FFT). The method allows a significant reduction in the number of computations as compared to ordinary Discrete Cosine Transform (DCT) coding procedure. See the Specification at pages 8-12.

For example, the method of claim 1 for coding audio data comprising a sequence of digital audio input samples includes the steps of:

i) multiplying the sequence of digital audio input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients.

Support for independent claim 1 can be found in claim 1 as originally filed, and on page 2, line 25, through page 3, line 10. More detail in the form of an example embodiment for performing the method is provided in Figure 3 and in the description on pages 8-12. Figure 3 is reproduced and discussed below. Applicants note that an International Preliminary Examination Report with Article 34 Amendments correcting an error in Figure 3 was filed with the U.S. National Phase Application. Figure 3 as reproduced below is the corrected Figure 3. Upon

remand to the Examiner, Applicants will verify that the correction was entered and, if necessary, make an appropriate amendment to correct Figure 3.

The embodiment of claim 6 additionally includes "wherein step i) comprises multiplying the sequence of digital audio input samples *x[n]* by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where: *x[n]* are the input sequence audio samples; *N* is the number of input sequence audio samples; and $n = 0,....,N-1$". Support for the embodiment of claim 6 can be found in claim 6 as originally filed, and in the form of an example embodiment employing the method provided in Figure 3 (see the illustrated input sequence and pre-multiplication) and the description on pages 7-12.

The embodiment of claim 8 additionally includes "wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r}\, cos(\pi(k+1/2)/N) - g_{k,i}\, sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r}\, cos(\pi(k+1/2)/N) + g_{k,i}\, sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively." Support for the embodiment of claim 8 can be found in claim 8 as originally filed, and in the form of an example embodiment employing the method provided in Figure 3 (see the illustrated post-multiplication block) and the description on pages 7-12.

The embodiment of claim 9 additionally includes "steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1\, cos(\pi(2k+1)/4) - T_2\, sin(\pi(2k+1)/4)$$

where Xk are the audio coded frequency domain coefficients; and $cos(\pi(2k+1)/4)$ and $sin(\pi(2k+1)/4)$ are the third trigonometric function factors. Support for the embodiment of claim 9 can be found in claim 9 as originally filed, and in the form of an example embodiment employing the method provided in Figure 3 (see the illustrated post-multiplication block) and the description on pages 7-12.

In another example, the embodiment of a method of claim 17 for coding audio data includes the steps of:

obtaining at least one input sequence of digital audio samples;

pre processing the input sequence samples including applying a pre multiplication factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence utilizing a fast Fourier transform; and

post processing the sequence of transform coefficients including applying first post multiplication factors to the real and imaginary coefficient components, differencing and combining the post multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples. Support for independent claim 17 can be found in claim 17 as originally filed, and on page 3, line 25, through page 4, line 5. More detail in the form of an example embodiment employing the method is provided in Figure 3 and in the description on pages 8-12.

Figure 3 of the specification, reproduced below for convenience, is a functional block diagram of an embodiment of a system that can be used to obtain the MDCT for a single audio channel.[1]

[1] As noted above, Figure 3 as it appears in the application contains an error, which Applicants will correct upon remand to the Examiner. Figure 3 as reproduced above is the corrected Figure 3.

Fast Modified Discrete Cosine Transform (single channel)

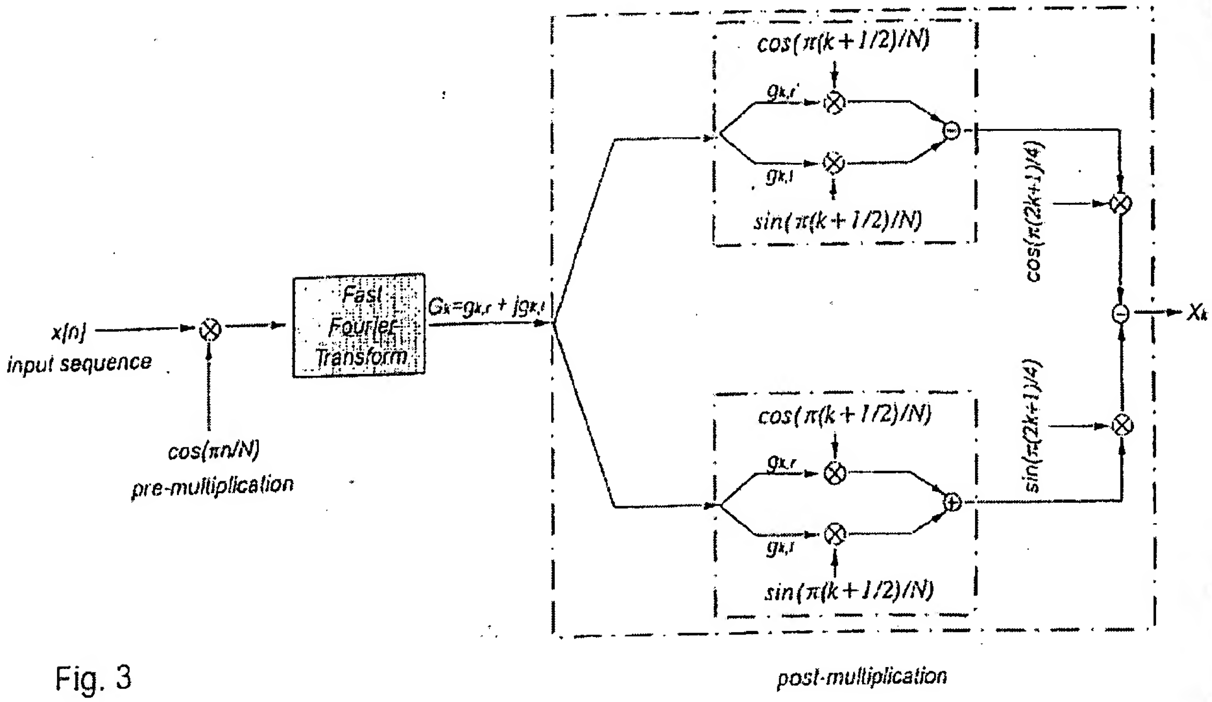

Fig. 3

The method can also be applied to significantly reduce the number of computations in a two-channel system. See pages 12-15 of the Specification.

For example, the embodiment of the method of claim 10 for coding audio data includes the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

processing the complex sample sequence by multiplying the input sequence samples by a first trigonometric function;

determining a Fourier transform coefficient sequence;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

Support for independent claim 10 can be found in claim 10 as originally filed, and on page 3, lines 12-23. More detail in the form of an example embodiment employing the method is provided in Figure 4 and in the description on pages 12-15, which incorporate by reference portions of the description of the single channel FFT illustrated in Figure 3 and the description on pages 8-12. Figure 4 is reproduced and discussed in more detail below.

The embodiment of claim 12 further includes "wherein the complex sample sequence is processed by multiplying the input sequence samples $z[n]$ by a first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate an intermediate sample sequence, where: $z[n] = x[n] + jy[n]$ is the complex sample sequence; $x[n]$ is the first sequence of digital audio samples; $y[n]$ is the second sequence of digital audio samples; N is the number of input sequence audio samples in each sequence; $n = 0,\ldots,N-1$; and j is the complex constant." Support for the embodiment of claim 12 can be found in claim 12 as originally filed, and in the form of an example embodiment provided in Figure 4 (see the pre-multiplication) and in the description on pages 12-15.

In another example, the embodiment of a method of claim 24 for coding audio data includes the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence utilizing a fast Fourier transform; and

post processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence

of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

Support for independent claim 24 can be found in claim 24 as originally filed, and on page 4, lines 7-26. More detail in the form of an example embodiment employing the method is provided in Figure 4 and in the description on pages 12-15, which incorporate by reference portions of the description of the single channel FFT illustrated in Figure 3 and the description on pages 8-12.

In another example, the embodiment of a method of claim 27 for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples x[n], y[n] corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence z[n], where z[n] = x[n] + jy[n];

pre processing the complex input sequence samples including applying a pre-multiplication factor cos(pn/N) + jsin(pn/N) to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and n = 0,....,(N 1);

transforming the modified complex input sequence samples into a complex transform coefficient sequence Zk utilizing a fast Fourier transform, wherein k = 0,....,(N12 1); and

post processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels Xk, Yk according to:

Gk = (Zk + Z*N-k-1)/2 k=0...N/2-1

G'k = (Zk – Z*N-k-1)/2j k=0...N/2-1

Xk = cosγ * (gk,rcos(p(k+1/2)/N)–gk,isin(p(k+1/2)/N))

– sinγ * (gk,rsin(p(k+1/2)/N)+gk,icos(p(k+1/2)/N))

$Yk = \cos\gamma * (g'k,r\cos(p(k+1/2)/N) - g'k,i\sin(p(k+1/2)/N))$

$- \sin\gamma * (g'k,r\sin(p(k+1/2)/N) + g'k,i\cos(p(k+1/2)/N))$

where

Gk is a transform coefficient sequence for the first channel;

G'k is a transform coefficient sequence for the second channel;

gk,r and gk,i are the real and imaginary transform coefficient components of Gk;

g'k,r and g'k,i are the real and imaginary transform coefficient components of G'k;

Z*N-k-1 is the complex conjugate of ZN-k-1; and

$\gamma(k) = p(2k+1)/4$.

Support for independent claim 27 can be found in claim 27 as originally filed, and on page 4, line 28 through page 5, line 20. More detail in the form of an example embodiment employing the method is provided in Figure 4 and in the description on pages 12-15, which incorporate by reference portions of the description of the single channel FFT illustrated in Figure 3 and the description on pages 8-12.

Figure 4 of the specification, reproduced below for convenience, is a functional block diagram of an embodiment of a system that can be used to obtain the MDCT for a pair of audio channels.

Combined Fast Modified Discrete Cosine Transform (two channels)

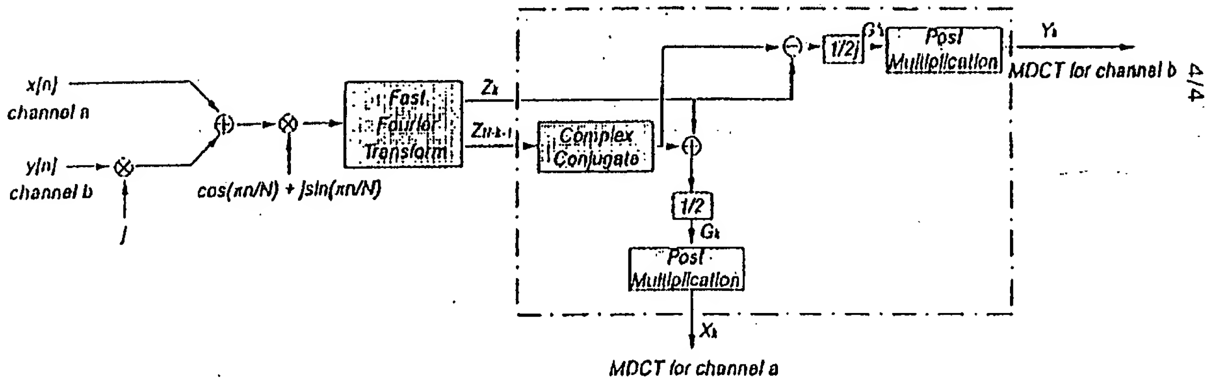

Fig. 4

In some embodiments, the method is applied when the pair of audio channels has the same selected transform length. See pages 12-15 of the Specification. In some embodiments, the method is applied by combining transforms of different lengths. See pages 15-16 of the Specification. In some embodiments, a windowing function is combined with a preprocessing stage to the transform, to further decrease the computational requirements. See pages 16-17 of the Specification.

Some embodiments are directed to apparatuses and systems for coding an input sequence of digital audio samples. See, for example, the embodiments illustrated in Figures 3 and 4 above. In one example embodiment, claim 28 is directed to an apparatus for encoding an input sequence of digital audio signals, the apparatus comprising:

a pre-transform processor to process the input sequence samples by applying a pre-multiplication factor to obtain modified input sequence samples;

a transform processor to apply a fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

a post-transform processor to process the sequence of transform coefficients by applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of audio coded frequency domain coefficients representing the input sequence of digital audio samples.

Support for independent claim 28 can be found in Figures 3 and 4 in the description on pages 8-15. Applicants note that Figure 3 illustrates an example embodiment having a pre-transform processor (illustrated as a multiplier and labeled pre-multiplication), a transform processor (illustrated as a fast Fourier Transform block), and a post transform processor (illustrated as a post-transform block).

The embodiment of claim 37 further includes the input sequence of digital audio samples comprising first and second sequences of digital audio samples from first and second audio channels;

the pre transform processor processing the first and second sequences of digital audio samples by combining the first and second sequences of digital audio samples into a single complex sample sequence;

the transform processor applying the fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

the post transform processor processing the transform coefficient sequence to thereby generate first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from the fast Fourier transform coefficient sequence, the post transform processor further processing each of the first and second transform coefficient sequences to thereby generate audio coded frequency domain coefficients so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels. Support for claim 37 can be found in Figures 3 and 4 in the description on pages 8-15. Applicants note that Figure 4 illustrates an example embodiment having a pre-transform processor combining first and second sequences into a complex sample sequence (illustrated as multiplier multiplying a channel by the complex constant j, a summer, and a multiplier), a

transform processor (illustrated as a fast Fourier Transform block), and a post transform processor (illustrated as a post-transform block).

The embodiment of claim 38 further includes the pre transform processor determining a transform length for each of the channels and pairing the channels according to their determined transform length, the coding the audio samples of first and second channels in each pair being performed based on the determined transform length. Support for claim 38 can be found in the description on pages 15-16.

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

1. Claims 1-9 and 17-23 were rejected under 35 U.S.C. § 103(a) as obvious over U.S. Patent No. 5,479,562 ("Fielder"). Applicants respectfully traverse the Examiner's contention that claims 1-9 and 17-23 are obvious over Fielder, and request that this ground for rejection be reviewed on appeal.

2. Claims 10-13, 16, and 24-27 were rejected under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis, et al, <u>Digital Signal Processing, principles, algorithms and applications</u> (3d ed. 1996) ("Proakis"). Applicants respectfully traverse the Examiner's contention that claims 10-13, 16 and 24-27 are obvious over Fielder in view of Proakis, and request that this ground for rejection be reviewed on appeal.

3. Claims 14-15 and 28-39 were rejected under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis and U.S. Patent No. 6,304,847 ("Jhung"). Applicants respectfully traverse the Examiner's contention that claims 10-13, 16 and 24-27 are obvious over Fielder in view of Proakis and Jhung, and request that this ground for rejection be reviewed on appeal.

VII. ARGUMENT

A. Introduction

The Federal Circuit has held many times that the Examiner must provide objective evidence of a motivation for combining the teachings of cited references in the manner claimed. *E.g., In re Sang-Su Lee*, 277 F.3d 1338, 1343; 61 USPQ2d 1430, 1433 (Fed. Cir. 2002). Further, "this factual question of motivation is material to patentability, and could not be resolved on subjective belief and unknown authority." *Id.* at 277 F.3d 1343-1344; 61 USPQ2d 1433.

Moreover, "the mere fact that the prior art may be modified in the manner suggested by the Examiner does not make the modification obvious unless the prior art suggested the desirability of the modification." *In re Fritch*, 972 F.2d 1260, 1266; 23 USPQ2d 1780, 1783-84.

The Examiner initially bears the burden of establishing a *prima facie* case of obviousness. *In re Bell*, 26 U.S.P.Q.2d 1529 (Fed. Cir. 1993); *In re Oetiker*, 977 F.2d 1443, 1445, 24 U.S.P.Q.2d 1443, 1444 (Fed. Cir. 1992); *In re Piasecki*, 745 F.2d 1468, 1472, 223 U.S.P.Q. 785, 788 (Fed. Cir. 1984); *MPEP* § 2142. An Applicant may attack an obviousness rejection by showing that the Examiner has failed to properly establish a *prima facie* case or by presenting evidence tending to support a conclusion of non-obviousness. *In re Fritch*, 972 F.2d at 1265.

In order for an examiner to establish a *prima facie* case that an invention, as defined by a claim at issue, is obvious the examiner must: (1) show some suggestion or motivation, either in the references themselves or in the knowledge generally available to one of ordinary skill in the art, to modify the reference or combine the reference teachings; (2) there must be a reasonable expectation of success; and (3) the prior art reference (or the combined references) must teach or suggest all the claim limitations. MPEP § 2142. "The teaching or suggestion to make the claimed combination and the reasonable expectation of success must both be found in the prior art, not in applicant's disclosure." MPEP § 2143. The level of skill in the art cannot be relied upon to provide the suggestion to combine the references. MPEP § 2143.01 (citing *Al-Site Corp. v. VSI Int'l Inc.*, 174 F.3d 1308, 50 U.S.P.Q.2d 1161 (Fed. Cir. 1999). The mere fact that the references <u>can</u> be combined or modified does not render the resultant combination obvious unless the prior art also suggests the desirability of the combination. MPEP § 2143.01 (citing *In re Mills*, 916 F.2d 680, 16 U.S.P.Q. 2d 1430 (Fed. Cir. 1990).

Moreover, a reference must be viewed as a whole, including portions that would lead away from the claimed invention. MPEP § 2141.03 (citing *W.L. Gore* & Assoc., *Inc. v. Garlock, Inc*, 721 F.2d 1540, 220 U.S.P.Q. 303 (Fed. Cir. 1983). If the proposed modification would change the principles of operation of the prior art invention being modified, then the teachings of the references are not sufficient to render the claims *prima facie* obvious. MPEP § 2143.01 (citing *In re Ratti*, 270 F.2d 810, 123 U.S.P.Q. 349 (CCPA 1959).

Here, the thrust of the Examiner's position is that the references ***could*** be modified to satisfy the claim limitations. The Examiner even goes so far as to provide elaborate proofs that

this is possible. The Examiner's proofs, however, illustrate why the Examiner is unable to make a prima facie case of obviousness. In each proof, the Examiner is forced to rely on "mathematical reasoning" for including certain intermediate steps in the proof. The Examiner cannot point to the references because the recited steps are not taught or suggested by the references.

Here, the Examiner admits that all of the recited steps are not disclosed by the cited references and improperly relies on the level of skill in the art to provide the motivation to combine (and further modify) the references. Merely showing the references could be combined (and further modified) so as to achieve the recited limitations is not enough to establish obviousness because it is not evidence that the cited references ***taught or suggested*** the recited limitations. Accordingly, the Examiner has not established a *prima facie* case of obviousness and the claims are allowable.

B. The Examiner Has Failed to Establish a Prima Facie Case That Fielder Renders Claims 1-9 and 17-23 Obvious

The Examiner rejected claims 1-9 and 17-23 under 35 U.S.C. § 103(a) as obvious over U.S. Patent No. 5,479,562 issued to Fielder, et al. The Examiner did not identify a second reference. Applicants respectfully traverse the Examiner's contention that Fielder renders claims 1-9 and 17-23 obvious and further submit that Fielder is not an appropriate primary reference.

The steps recited in claim 1, as amended, include:

> iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;
>
> iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

> v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients."

The Examiner states that "Fielder does not expressly disclose every intermediate result or step in the processing." The Examiner argues that "those intermediate results or steps of the processing are obvious to one skilled in the art to be obtained through a simple mathematical reasoning, because Fielder teaches a starting equation (24), condition equations (6 and 25), intermediate equations (26, 27 and some part of 28), and final result equation (28)." Final Office Action, paragraph 8, page 7. The Examiner points to no portion of Fielder suggesting that the claimed intermediate steps should be derived. In fact, the Examiner admits that there are "alternate computations for the process" and refers to ***examples*** of how Fielder ***could*** employ the claimed or "equivalent" intermediate steps. *See, e.g.* Final Office Action, paragraph 8, page 6; discussion of claim 6, on pages 8 and 9. Further, one of the references cited by the Examiner, Proakis, et al, <u>Digital Signal Processing, principles, algorithms and applications</u> (3d ed. 1996) ("Proakis"), discussed in more detail below, clearly indicates that there are choices among FFT algorithms:

> In conclusion, we have presented several important considerations in the implementation of FFT algorithms. Advances in digital signal processing technology, in hardware and software, will continue to influence ***the choice among FFT algorithms for various practical applications.***

Proakis, page 475, fourth paragraph (emphasis added).

Thus, Applicants respectfully traverse the Examiner's contention that Fielder renders claims 1-9 and 17-23 obvious and submit that the Examiner has failed to establish a prima facie case of obviousness. *See* MPEP § 2143.01 (The mere fact that references can be combined or modified does not render the resultant combination obvious unless the prior art also suggests the desirability of the combination."); MPEP § 2112 ("The mere fact that a certain result or characteristic <u>may</u> occur or be present in the prior art is not sufficient to establish the inherency of that result or characteristic.") (emphasis in original).

1. Claims 1-9 are not rendered obvious by Fielder

Returning to the language of the claims, there is no teaching or suggestion in Fielder to: "iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient; iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients", as recited in claim 1. The Examiner points in a conclusory fashion to Equation 28 of Fielder and a "post multiply step" in Fielder as rendering obvious all three of the recited steps. The Examiner fails to indicate how Fielder discloses or teaches: generating "an addition stream coefficient"; generating "a subtraction stream coefficient"; "multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors"; or "subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients" as claimed. The Examiner's whole argument is that the particularly claimed intermediate steps to generate coded frequency domain coefficients were mathematically possible, not that Fielder actually disclosed or inherently used the particular intermediate steps claimed. The Examiner has not cited Fielder as an anticipating reference. The Examiner has not even shown that Fielder generated the same coded frequency domain coefficients. To establish inherency, the extrinsic evidence must make clear that the missing descriptive matter is necessarily present in the thing described in the reference, and that it would be so recognized by persons of ordinary skill. MPEP § 2112. The mere fact that the claimed intermediate steps ***could*** have been used is not sufficient.

Claims 2-9 depend from claim 1. Accordingly, Applicants submit that claims 1-9 are not rendered obvious by Fielder and that the Examiner has failed to establish a *prima facie* case of obviousness.

2. Claim 6 Recites Additional Limitations That Are Not Rendered Obvious by Fielder

In addition, with regard to claim 6, the Examiner states that "Fielder does not expressly disclose" additional limitations in the claim. Final Office Action, page 8. Specifically, claim 6 recites "wherein step i) comprises multiplying the sequence of digital audio input samples $x[n]$ by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where: $x[n]$ are the input sequence audio samples; N is the number of input sequence audio samples; and $n = 0,....,N-1$." The Examiner uses the same faulty reasoning applied to claim 1 (and relied upon by the Examiner in every rejection) – namely that Fielder ***could*** have preformed the recited steps in the recited manner, not that Fielder ***actually*** disclosed or suggested performing the recited steps in the recited manner. Accordingly, Applicants respectfully submit that claim 6 is not rendered obvious by Fielder for the additional reason that the limitations set forth above are not taught or suggested by Fielder and there is no motivation in Fielder for the combination.

3. Claim 8 Recites Additional Limitations That Are Not Rendered Obvious by Fielder

With regard to claim 8, the Examiner states that "Fielder does not expressly disclose" additional limitations in the claim. Final Office Action, page 9. Specifically, claim 8 recites "wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r}\, cos(\pi(k+1/2)/N) - g_{k,i}\, sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r}\, cos(\pi(k+1/2)/N) + g_{k,i}\, sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively." The Examiner uses the same faulty reasoning applied to claim 1 (and relied upon by the Examiner in every rejection) – namely that Fielder ***could*** have preformed the recited steps in the recited manner, not that Fielder ***actually*** disclosed or suggested performing the recited steps in the recited manner. Accordingly, Applicants respectfully submit that claim 8 is not rendered obvious by Fielder for the additional reason that the limitations set forth above are not taught or suggested by Fielder and there is no motivation in Fielder for the combination.

4. Claim 9 Recites Additional Limitations That Are Not Rendered Obvious by Fielder

With regard to claim 9, the Examiner states that "Fielder does not expressly disclose" additional limitations in the claim. Final Office Action, page 10. Specifically, claim 9 recites "steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1 \cos(\pi(2k+1)/4) - T_2 \sin(\pi(2k+1)/4)$$

where Xk are the audio coded frequency domain coefficients; and $\cos(\pi(2k+1)/4)$ and $\sin(\pi(2k+1)/4)$ are the third trigonometric function factors." The Examiner uses the same faulty reasoning applied to claim 1 (and relied upon by the Examiner in every rejection) – namely that Fielder ***could*** have preformed the recited steps in the recited manner, not that Fielder ***actually*** disclosed or suggested performing the recited steps in the recited manner. Accordingly, Applicants respectfully submit that claim 9 is not rendered obvious by Fielder for the additional reason that the limitations set forth above are not taught or suggested by Fielder and there is no motivation in Fielder for the combination.

5. Claims 17-23 Are Not Rendered Obvious by Fielder

Claim 17 recites: "post-processing the sequence of transform coefficients including applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples." The Examiner relies on the reasoning applied to claim 1. The Examiner makes no attempt to identify what portion of Fielder teaches or suggests "applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing" as recited.

Claims 18-23 depend from claim 17. Accordingly, Applicants submit that claims 17-23 are not rendered obvious by Fielder and that the Examiner has failed to establish a *prima facie* case of obviousness.

C. The Examiner Has Failed to Establish a Prima Facie Case That Fielder in View of Proakis Renders Claims 10-13, 16 and 24-27 Obvious

The Examiner rejected claims 10-13, 16 and 24-27 under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis, et al, Digital Signal Processing, principles, algorithms and applications (3d ed. 1996) ("Proakis"). Applicants respectfully traverse the Examiner's contention that Fielder in view of Proakis renders claims 10-13, 16 and 24-27 obvious and submit that the Examiner has failed to establish a *prima facie* case of obviousness.

Applicants respectfully submit that Fielder is not an appropriate primary reference and further that modifying Fielder as suggested by the Examiner would improperly change the function and principles of operation of Fielder.

Claim 10 as amended recites: "combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence; processing the complex sample sequence by multiplying the input sequence samples by a first trigonometric function; determining a Fourier transform coefficient sequence; generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels." Similarly, claim 24 recites "for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients."

The Examiner states that "Fielder does not expressly disclose all detailed intermediate steps." The Examiner does not argue the specific intermediate steps missing from Fielder are taught by Proakis, and points to no specific language in Proakis disclosing or suggesting the specific recited limitations. The Examiner points to equations 4.3.37 and 5.2.31, Tables 4.4 and

5.1 and Figure 4.29, which merely describe properties of the Discrete Fourier Transform, and do not disclose or suggest the claimed intermediate steps.

Further, both Fielder and Proakis teach away from the claimed invention. Fielder teaches using two FFT processes in two channel systems: "In two-channel systems, signal sample blocks from each of the two channels are transformed by ***two*** FFT processes into a DCT_1/DCT_2 block pair." See Column 36, lines 38-41 (emphasis added). Thus, Fielder does not teach or suggest using ***a*** Fourier transform coefficient sequence to generate first and second transform coefficient sequences as the Examiner suggests. Fielder instead teaches away from the claimed invention by teaching that in a two-channel system, two FFT processes should be used. Proakis assumes as a condition that the input sequences to the transform are real-valued sequences. See page 475-76. Thus, Proakis assumes there is no need for the claimed intermediate steps. Claims 11-13 and 16 depend from claim 10 and claims 25-26 depend from claim 24. The Examiner argues that Applicants are ignoring another embodiment (apparently in Proakis), but points to the very embodiment Applicants discussed, which assumes the input sequences are real-valued sequences. Accordingly, Applicants respectfully submit that claims 10-13, 16, and 25-26 are not render obvious by Fielder in view of Proakis, and that the Examiner has failed to establish a *prima facie* case of obviousness.

1. Claim 12 Recites Additional Limitations That Are Not Rendered Obvious by Fielder

In addition, with regard to claim 12, the Examiner incorporates the argument made with regard to claim 6, which, like all of the Examiner's arguments, is based on the flawed premise that obviousness can be shown by establishing that a reference "may" be modified to include a recited claim limitation. Specifically, claim 12 recites: "wherein the complex sample sequence is processed by multiplying the input sequence samples $z[n]$ by a first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate an intermediate sample sequence, where: $z[n] = x[n] + jy[n]$ is the complex sample sequence; $x[n]$ is the first sequence of digital audio samples; $y[n]$ is the second sequence of digital audio samples; N is the number of input sequence audio samples in each sequence; $n = 0,....,N-1$; and j is the complex constant." This is not taught or suggested by Fielder, as the Examiner admits, and the Examiner does not contend this is taught or suggested by Proakis. Accordingly, Applicants respectfully submit that claim 12 is not rendered obvious

by Fielder in view of Proakis, and that the Examiner has failed to make a *prima facie* showing of obviousness with regard to claim 12 for this additional reason.

2. Claim 27 Recites Additional Limitations That Are Not Rendered Obvious by Fielder

Claim 27 recites "obtaining first and second input sequences of digital audio samples *x[n]*, *y[n]* corresponding to respective first and second audio channels; combining the first and second input sequences of digital audio samples into a single complex input sample sequence *z[n]*, where $z[n] = x[n] + jy[n]$; pre-processing the complex input sequence samples including applying a pre-multiplication factor $cos(\pi n/N) + jsin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,....,(N-1)$; transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k utilizing a fast Fourier transform, wherein $k = 0,....,(N12- 1)$; and post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0...N/2-1$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0...N/2-1$$

$$X_k = \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N))$$
$$- \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N))$$

$$Y_k = \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N))$$
$$- \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N))$$

where

G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k) = \pi(2k+1)/4$.

The Examiner again relies on an assertion that Fielder in view of Proakis and mathematical reasoning "could" be modified to incorporate the additional limitations of claim

27. As set forth above, establishing that a reference "could" be modified to incorporate the limitations of a claim does not establish that the claimed invention is obvious. There is no "suggestion" in the cited reference (or in "mathematical reasoning") to modify Fielder as suggested by the Examiner. Further, as discussed above, Fielder and Proakis teach away from claimed invention. Fielder teaches that in a two-channel system, two FFT processes should be used. Proakis assumes as a condition that the input sequences to the transform are real-valued sequences. Accordingly, Applicants respectfully submit that claim 27 is not rendered obvious by Fielder in view of Proakis, and that the Examiner has failed to make a *prima facie* showing of obviousness with regard to claim 27.

D. The Examiner Has Failed to Establish a Prima Facie Case That Fielder in View of Proakis and Jhung Renders Claims 14-15 and 28-39 Obvious

The Examiner rejected claims 14-15 and 28-39 under 35 U.S.C. § 103(a) as obvious over Fielder in view of Proakis and U.S. Patent No. 6,304,847 issued to Jhung. Claims 14 and 15 depend from claim 10, which, as discussed above, is not rendered obvious by the combination of Fielder and Proakis. The Examiner does not contend that Jhung teaches or suggests the claimed intermediate steps missing from Fielder and Proakis.

1. Claims 28-39 Are Not Rendered Obvious By Fielder In View of Proakis and Jhung

Independent claim 28 recites "a transform processor to apply a fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and a post-transform processor to process the sequence of transform coefficients by applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of audio coded frequency domain coefficients representing the input sequence of digital audio samples." Claims 29-39 depend from claim 28. The Examiner merely points to the arguments made with respect to claims 1, 10 and 14. The claimed "post-transform processor to process the sequence of transform coefficients by applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second

post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of audio coded frequency domain coefficients representing the input sequence of digital audio samples" are not taught or suggested by Fielder, alone or in combination with Proakis and the Examiner does not contend they are taught or suggested by Jhung. Accordingly, Applicants respectfully submit that claims 14-15 and 28-39 are not rendered obvious by Fielder in view of Proakis and Jhung, and that the Examiner has failed to establish a *prima facie* case of obviousness.

2. Claims 37 and 38 Contain Additional Limitations That Are Not Rendered Obvious By Fielder In View of Proakis and Jhung

In addition, claim 37, which depends from claim 28, recites: "wherein the input sequence of digital audio samples comprises first and second sequences of digital audio samples from first and second audio channels; the pre-transform processor processing the first and second sequences of digital audio samples by combining the first and second sequences of digital audio samples into a single complex sample sequence." For reasons similar to those set forth above with regard to claim 10, claims 37 and 38, which depends from claim 37, are not rendered obvious by Fielder in view of Proakis and Jhung, and the Examiner has failed to establish a *prima facie* case of obviousness with respect to claims 37 and 38 for this additional reason.

VIII. CLAIMS APPENDIX

A copy of the claims as currently pending is attached hereto as Appendix A.

IX. APPENDIX OF EVIDENCE

The Final Office Action mailed on December 17, 2004, and referred to herein is attached hereto as Appendix B.

A copy of a reference cited by the Examiner and referred to herein is attached hereto as Appendix C. The reference was cited in an Office Action mailed on April 23, 2004.

This application is a conversion of PCT Application No. PCT/SG98/00014 filed February 21, 1998, into a U.S. National Application, which is referenced herein. Accordingly, a copy of PCT Application No. PCT/SG98/00014 filed February 21, 1998, is attached hereto as Appendix D.

A copy of an International Preliminary Examination Report including an Article 34 amendment correcting Figure 3 was filed with the U.S. National Application. The corrected Figure

3 is referenced herein. Accordingly, a copy of the International Preliminary Examination Report amending Figure 3 is attached hereto as Appendix E.

X. RELATED PROCEEDINGS APPENDIX

There are no related proceedings, and thus no related proceedings are included in a Related Proceedings Appendix. The Notification of Non-Complaint Appeal Brief, however, requested a Related Proceedings Appendix. Accordingly, Applicants have included a Related Proceedings Appendix attached hereto as Appendix F, which indicates that there are no related proceedings.

XI. CONCLUSION

The Examiner has failed to establish a *prima facie* case that the claims are rendered obvious by Fielder, whether considered alone or in combination with Proakis and/or Jhung. Moreover, with respect to claims 10-17 and 27, the cited references teach away from the claimed invention. Accordingly, allowance of the claims is respectfully requested.

Respectfully submitted,

Seed Intellectual Property Law Group PLLC

Timothy L. Boller
Registration No. 47,435

TLB:rr
Enclosures:
Appendix A
Appendix B
Appendix C
Appendix D
Appendix E
Appendix F

701 Fifth Avenue, Suite 6300
Seattle, Washington 98104-7092
Phone: (206) 622-4900
Fax: (206) 682-6031

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APPENDIX A

CLAIMS INVOLVED IN THE APPEAL

1. A method for coding audio data comprising a sequence of digital audio input samples, including the steps of:

i) multiplying the sequence of digital audio input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients.

2. A method for coding audio data as claimed in claim 1, wherein the audio coded frequency domain coefficients comprise modified discrete cosine transform coefficients.

3. A method for coding audio data as claimed in claim 1, wherein the first trigonometric function factor for each digital audio input sample is a function of a sequence position of the digital audio input sample and the number of samples in the sequence.

4. A method for coding audio data as claimed in claim 1, wherein the respective second trigonometric function factors for each transform coefficient in the sequence are respective functions of the transform coefficient sequence position and the number of coefficients in the sequence.

5. A method for coding audio data as claimed in claim 1, wherein the respective third trigonometric function factors are respective functions of the transform coefficient sequence position.

6. A method for coding audio data as claimed in claim 1, wherein step i) comprises multiplying the sequence of digital audio input samples $x[n]$ by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where:

$x[n]$ are the input sequence audio samples;

N is the number of input sequence audio samples; and

$n = 0,....,N-1$.

7. A method for coding audio data as claimed in claim 1, wherein step ii) comprises computing the fast Fourier transform of the intermediate sample sequence so as to generate said transform coefficient sequence $G_k = g_{k,r} + jg_{k,i}$, where:

G_k is the transform coefficient sequence;

$g_{k,r}$ are the real transform coefficient components;

$g_{k,i}$ are the imaginary transform coefficient components; and

$k = 0,....,(N/2-1)$.

8. A method for coding audio data as claimed in claim 1, wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r}\cos(\pi(k+1/2)/N) + g_{k,i}\sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively.

9. A method for coding audio data as claimed in claim 1, wherein steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1\cos(\pi(2k+1)/4) - T_2\sin(\pi(2k+1)/4)$$

where Xk are the audio coded frequency domain coefficients; and

$\cos(\pi(2k+1)/4)$ and $\sin(\pi(2k+1)/4)$ are the third trigonometric function factors.

10. A method for coding audio data, including the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

processing the complex sample sequence by multiplying the input sequence samples by a first trigonometric function;

determining a Fourier transform coefficient sequence;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

11. A method for coding audio data as claimed in claim 10, wherein the step of generating first and second transform coefficient sequences comprises, for each corresponding coefficient in the first and second transform coefficient sequences, selecting first and second transform coefficients from said Fourier transform coefficient sequence, determining a complex conjugate of said second transform coefficient, combining said first transform coefficient and said complex conjugate for said first transform coefficient sequence and differencing said first transform coefficient and said complex conjugate for said second transform coefficient sequence.

12. A method for coding audio data as claimed in claim 10, wherein the complex sample sequence is processed by multiplying the input sequence samples $z[n]$ by a first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate an intermediate sample sequence, where:

$z[n] = x[n] + jy[n]$ is the complex sample sequence;

$x[n]$ is the first sequence of digital audio samples;

$y[n]$ is the second sequence of digital audio samples;

N is the number of input sequence audio samples in each sequence;

$n = 0,\ldots,N\text{-}1$; and

j is the complex constant.

13. A method for coding audio data as claimed in claim 11, wherein said first and second transform coefficient sequences are generated according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j$$

where G_k is said first transform coefficient sequence;

G'_k is said second transform coefficient sequence;

N is the number of input sequence audio samples;

$k = 0,\ldots,(N/2\text{-}1)$;

Z_k is said first transform coefficient;

Z^*_{N-k-1} is the complex conjugate of said second transform coefficient; and

j is the complex constant.

14. A method for coding audio data as claimed in claim 10 further comprising examining said first and second sequences of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

15. A method for coding audio data comprising sequences of digital audio samples from a plurality of audio channels as defined in claim 10, further comprising determining a transform length for each of the channels, pairing the channels according to their determined transform length, and coding the audio samples of first and second channels in each pair according to the determined transform length.

16. A method for coding audio data as claimed in claim 10, including applying a windowing function in combination with multiplying the complex sample sequence by a first trigonometric function factor.

17. A method for coding audio data including the steps of:

obtaining at least one input sequence of digital audio samples;

pre-processing the input sequence samples including applying a pre-multiplication factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence utilizing a fast Fourier transform; and

post-processing the sequence of transform coefficients including applying first post-multiplication factors to the real and imaginary coefficient components, differencing and

combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

18. A method as claimed in claim 17, wherein the pre-multiplication factor, and first and second post-multiplication factors are trigonometric function factors.

19. A method as claimed in claim 17, wherein the pre-multiplication factor applied to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

20. A method as claimed in claim 17, wherein the first post-multiplication factors for each transform coefficient in the sequence are trigonometric functions of the transform coefficient sequence position and the number of coefficients in the sequence.

21. A method as claimed in claim 17, wherein the second post-multiplication factor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

22. A method as claimed in claim 17, wherein the pre-processing operations are performed on each sample in the input sequence individually.

23. A method as claimed in claim 17, wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

24. A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre-processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence utilizing a fast Fourier transform; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

25. A method as claimed in claim 24, wherein the pre-multiplication factor for each sample in the complex input sample sequence comprises a complex trigonometric function of the complex input sample sequence position and the number of samples in the sequence.

26. A method as claimed in claim 24, wherein the post-processing for each of the first and second channels includes applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and

imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

27. A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples $x[n]$, $y[n]$ corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence $z[n]$, where $z[n] = x[n] + jy[n]$;

pre-processing the complex input sequence samples including applying a pre-multiplication factor $\cos(\pi n/N) + j\sin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,\ldots.,(N-1)$;

transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k utilizing a fast Fourier transform, wherein $k = 0,\ldots.,(N12- 1)$; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0\ldots N/2-1$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0\ldots N/2-1$$

$$X_k = \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N))$$
$$- \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N))$$

$$Y_k = \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N))$$
$$- \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N))$$

where

G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k)$

$= \pi(2k+1)/4$.

28. An apparatus for coding an input sequence of digital audio samples comprising:

a pre-transform processor to process the input sequence samples by applying a pre-multiplication factor to obtain modified input sequence samples;

a transform processor to apply a fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

a post-transform processor to process the sequence of transform coefficients by applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of audio coded frequency domain coefficients representing the input sequence of digital audio samples.

29. The apparatus of claim 28 wherein the sequence of audio coded frequency domain coefficients are modified discrete cosine transform coefficients

30. The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor, and first and second post-multiplication factors applied by the post-transform processor are trigonometric function factors.

31. The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

32. The apparatus of claim 28 wherein the first post-multiplication factors applied by the post-transform processor for each transform coefficient in the sequence are trigonometric functions of the transform coefficient sequence position and the number of coefficients in the sequence.

33. The apparatus of claim 28 wherein the second post-multiplication factor applied by the post-transform processor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

34. The apparatus of claim 28 wherein the pre-multiplication factor applied by the pre-transform processor are applied to each sample in the input sequence individually.

35. The apparatus of claim 28 wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

36. The apparatus of claim 28 wherein the pre-transform processor is configured to analyze the input sequence of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

37. The apparatus of claim 28 wherein the input sequence of digital audio samples comprises first and second sequences of digital audio samples from first and second audio channels;

the pre-transform processor processing the first and second sequences of digital audio samples by combining the first and second sequences of digital audio samples into a single complex sample sequence;

the transform processor applying the fast Fourier transform to the modified input sequence samples to thereby generate a transform coefficient sequence having real and imaginary coefficient components; and

the post-transform processor processing the transform coefficient sequence to thereby generate first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from the fast Fourier transform coefficient sequence, the post-transform processor further processing each of the first and second transform coefficient sequences to thereby generate audio coded frequency domain coefficients so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

38. The apparatus of claim 37 wherein the pre-transform processor determines a transform length for each of the channels and pairs the channels according to their determined transform length, the coding the audio samples of first and second channels in each pair being performed based on the determined transform length.

39. A apparatus of claim 28, further comprising applying a windowing function in combination with the pre-multiplication factor.

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APPENDIX B

FINAL OFFICE ACTION MAILED DECEMBER 17, 2004

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UNITED STATES PATENT AND TRADEMARK OFFICE

OIPE JUL 10 2006 PATENT & TRADEMARK OFFICE

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/622,736 ✓	10/27/2000 ✓	Mohammed Javed Absar ✓	851663.413US ✓	2744

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EXAMINER
HAN, QI

ART UNIT	PAPER NUMBER
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Seed Intellectual Property Law Group
701 Fifth Avenue Suite 6300
Seattle, WA 98104-7092

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Please find below and/or attached an Office communication concerning this application or proceeding.

2 - month Response Due: FEB 17, 2005
3 - month Response Due: MAR 17, 2005
Notice of Appeal Due: JUN 17, 2005
(6 - month period ends) Will Go Aban
(3 - month extension of the time required)

FINAL REJECTION

ENTERED IN DOCKET RG

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JUL 1 0 2006
PATENT & TRADEMARK OFFICE
IAP69

Office Action Summary	Application No.	Applicant(s)	
	09/622,736	ABSAR ET AL.	
	Examiner	Art Unit	
	Qi Han	2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 27 August 2004.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-39 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-39 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 08/27/2004.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

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DETAILED ACTION

1. The text of those sections of Title 35, U.S. Code not included in this action can be found in a prior Office action.

Information Disclosure Statement

2. The references listed in the Information Disclosure Statement submitted on 08/27/2004 have been considered by the examiner (see attached PTO-1449).

Response to Amendments

3. This communication is responsive to the applicant's amendment dated 08/23/2004. Applicant amended claims 1, 3, 6, 10 and 27 (see the amendment: pages 2-8 and 13).

4. Examiner withdraws the disclosure objections a-f, because applicant made correction(s) and/or amendment(s).

5. Examiner withdraws the claim objection regarding claim 27, because applicant made correction(s) and/or amendment(s).

6. Examiner withdraws the rejection of claims 1-2,4-5 and 7-8 under 35 USC 101, because applicant made correction(s) and/or amendment(s).

Response to Arguments

7. Applicant's arguments with respect to the independent claims 1-39 have been fully considered but are not persuasive.

In response to applicant's arguments regarding claims 1-9 and 17-23 that "the examiner point to no portion of Fielder suggesting that the claimed intermediate steps should be derived", "Nor does the establish that the steps are inherent", "the examiner fails to indicate how Fielder discloses or teaches: generating ... as claimed" and "the examiner makes no attempt to identify what portion of Fielder teaches or suggests applying first post-multiplication factor...as claimed" (the amendment, page 18, paragraph 5 to page 20 paragraph 2), the examiner respectfully disagrees with applicant and has a different view of the prior art teachings and the claim interpretations. It is should pointed out that the argument says "the examiner concedes that Fielder does not disclose the claimed intermediate steps", which twists the examiner's words of "Fielder **does not expressly disclose all intermediate results or steps** in processing", which means some intermediate results or steps have been disclosed and some of them may be implicated by or derived from the disclosure (see the detail in the claim rejection in the office action). By reviewing the claim limitations, specification, and the rejection under 35USC 103 in previous office action, it is noted that the recited claim limitations are based on the mathematical reasoning steps disclosed in the specification, so that examiner's rejection follows the same manner and covers all the limitations as claimed (see detail in the rejection). Further, It is noted that the prior art starts with MDCT (equation 24) that is same as claimed, gives some intermediate processing steps or conditions (equations 25-27), and resulting an equivalent

expression (equation 28) (also see column 35, line 40 to column 36, lines 57), which can be derived to the claimed results by simple mathematical reasoning(s). It is also noted that mathematical equivalent expression can be interchangeable, and the intermediate steps or results either may be necessarily included or can be inherently derived, so that it is obvious to one skilled in the art to make the mathematical reasoning(s). For example, equation 28 can be simply derived to C(k)= R(k)cos (a+b) + Q(k)sin(a+b) = R(k) [cos (a)cos (b) - (sin(a)sin(b)] – Q(k)[sin(a)cos(b)+cos(a)sin(b)] =cos(b)[R(k) cos(a) - Q(k) sin(a)] – sin(b)[R(k) sin (a) + Q(k) cos(a)], which is read on the claimed limitation.

In response to applicant's arguments regarding claims 10-13,16 and 24-27 that "examiner does not argue the intermediate step missing from Fielder are taught by Proakis, and points to no specific language in Proakis disclosing or suggesting the recited limitations", "both Fielder and Proakis teaches away from the claimed invention" (the amendment, page 20, paragraph 3 to page 21 paragraph 2), the examiner respectfully disagrees with applicant and has a different view of the prior art teachings and the claim interpretations. Regarding "the intermediate steps" issue, the response is based on the same reason as stated above. Regarding "a Fourier transform" issue (page 21, paragraph 2), it is noted that applicant only takes one embodiment as his argument base, but not argue the other portion the prior art disclosure referred by the examiner (see detail in the claim rejection). In addition, by reviewing the claim rejection in previous office action, Proakis teaches symmetry properties of the discrete-time Fourier transform (page 290-291) that disclose the mathematical relationships between different time domain/frequency domain signal components, including even/odd, real/ image, and conjugate relationship between input and output signals (page 190, equation 4.3.37, Fig. 4,29, page 415, equation 5.231, table 4.4 and table

5.1), and efficient computation of the DFT of two real sequences (page 475-476) that can compute two real signal sequences in a complex-valued sequence by performing a single DFT (FFT). Furthermore, it is noted that the reference of Proakis is a textbook, which teaches the general mathematic properties of the discrete-time Fourier transform for various signal processing application. Therefore, Therefore, it would have been obvious to one of ordinary skill in the art to apply or combine the teachings of the symmetry properties of the Fourier transform for various signal processing application, including audio signal processing as claimed, for the purpose of enhancing the efficiency of the FFT algorithm (Proakis: page 475, paragraph 6).

Regarding applicant's arguments regarding claims 14-15 and 28-29 (the amendment: page 21, paragraphs 3-4), the response is based on the same reason as stated above (also see detail in the claim rejections).

Therefore, as stated above, examiner believes that the rejection is proper and the applicant's arguments are not persuasive. In addition, there are some changes in the rejection of this office action, for reflecting the applicant's arguments and correcting miner errors, without changing the prior art teachings.

Claim Rejections - 35 USC § 103

8. Claims 1-9 and 17-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over Fielder et al. (US 5,479,562) hereinafter referenced as Fielder.

As per **claim 1**, Fielder discloses a method and apparatus for encoding and decoding audio information (title), comprising:

i) multiplying the sequence of digital audio input samples with a first trigonometric function factor to generate an intermediate sample sequence (column 35, line 35 to column 36, line 8 and equation (26), 'premultiply step') ;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence (column 36, lines 9-19 and equation (27));

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficients (column 36, lines 20-35 and equation (28), 'a postmultiply step', wherein equation (28) has equivalent function and same result as the equation 16 in the specification, for example applying trigonometric equation cos(a+b)=cos(a)cos(b)-sin(a)sin(be) and sin(a+b)=sin(a)cos(b)+cos(a)sin(b) to equation (2), which produces an equivalent expression and is read on the claimed limitation);

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors (column 36, lines 20-35 and equation (28), 'a postmultiply step', wherein equation (28) has equivalent function and same result as the equation 16 in the specification, which reads on the claimed limitation); and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients (column 36, lines 20-35 and equation (28), 'a postmultiply step', wherein equation (28) has equivalent function and same result as the equation 16 in the specification, for example applying trigonometric equation cos(a+b) =

cos(a)cos(b)-sin(a)sin(be) and sin(a+b)=sin(a)cos(b)+cos(a)sin(b) to equation (2), which produces an equivalent expression and is read on the claimed limitation which reads on the claimed limitation);

Even though Fielder discloses some the intermediate results or steps, Fielder dos not expressly disclose every intermediate result or step in the processing. However, those intermediate results or steps of the processing are obvious to one skilled in the art to be obtained through a simple mathematical reasoning, because Fielder teaches a starting equitation (24), condition equations (6 and 25), intermediate equations (26, 27 and some part of 28), and final result equation (28), which correspond to Eq.1, Eq. 13 and Eq. 16 of the specification and can derive or inherently include other intermediate steps of using trigonometric functions, multiplying addition and subtraction processes as claimed. For example, equation 28 can be simply derived to C(k)= R(k)cos (a+b) + Q(k)sin(a+b) = R(k) [cos (a)cos (b) - (sin(a)sin(b)] – Q(k)[sin(a)cos(b)+cos(a)sin(b)] =cos(b)[R(k) cos(a) - Q(k) sin(a)] – sin(b)[R(k) sin (a) + Q(k) cos(a)], which equivalent to the claimed limitation. Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to compute Fielder' the equations by a simple mathematical reasoning, for the purpose of providing a complete computation algorithm to reach the final conclusion (equation 28) from a staring point (equation 24) (Fielder: column 35, lines 60-67).

As per **claim 2** (depending on claim 1), Fielder further discloses the audio coded frequency domain coefficients comprise modified discrete cosine transform coefficients (column 35, lines 33-59).

As per **claim 3** (depending on claim 1), Fielder further discloses that the first trigonometric function factor for each audio sample is a function of the audio sample sequence position (n) and the number (N) of samples in the sequence (column 36, eq. (26)).

As per **claim 4** (depending on claim 1), Fielder further discloses that the respective second trigonometric function factors for each transform coefficient in the sequence are respective functions of the transform coefficient sequence position (k) and the number (N) of coefficients in the sequence (column 36, eq. (28)).

As per **claim 5** (depending on claim 1), Fielder further discloses that the respective third trigonometric function factors are respective functions of the transform coefficient sequence position (k) (column 36, eq. (28) and column 18 eq. (6), angle of 2pi (k+ ½) m/N = 2pi (k+ ½)/4 + pi (k+ ½), where m=(N/2 +1)/2, so that a trigonometric function factor of sum of two angles can be expressed by a sum (two terms) of trigonometric function factors with individual angles, i.e. cos(a+b) = cos (a) cos(b) - sin(a)sin(b), therefore one of the trigonometric function factors can be read on the claimed limitation).

As per **claim 6** (depending on claim 1), Fielder does not expressly disclose that the step i) comprises multiplying the input sequence samples x[n] by the first trigonometric function factor cos(πn/N) to generate the intermediate sample sequence, where: x[n] are the input sequence audio samples; N is the number of input sequence audio samples. However, Fielder discloses multiplying the input sequence samples x[n] by cos(- πn/N) (column 36, eqs. (26) and (27), where exp(-j πn/N) = cos(- πn/N) + j sin (- πn/N)), which is symmetric to eq. 11 in the specification (page 10) and also corresponds to eq. 13 (specification: page 11), which suggests that computing intermediate result in eq. 13 by using factor exp(-j πn/N) has equivalent

functionality of computing intermediate result in eq. 12, since there is a conjugation relationship between them). Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder by providing multiplying the input sequence samples x[n] by an exchangeable a factor, either exp(j πn/N) or exp(-j πn/N), for the purpose of providing an alternative computation for the process.

As per **claim 7** (depending on claim 1), Fielder further discloses that step ii) comprises computing the fast Fourier transform of the intermediate sample sequence so as to generate said transform coefficient sequence Gk = gk,r + jgk,i, where: Gk is the transform coefficient sequence; gk,r are the real transform coefficient components; gk,i are the imaginary transform coefficient components; and k=O (N/2 - 1), (column 36, eqs. (27) and (28), where X*(k), R(k) and Q(k) correspond to Gk, gk,r and gk,i, respectively).

As per **claim 8** (depending on claim 1), Fielder does not expressly disclose that step iii) comprises determining the addition stream coefficients T2 and subtraction stream coefficients T, according to: T1 = gk,r cos (pi(k + 1/2)/N) - gk,i sin (pi(k + 1/2)/N); T2 =gk,r cos(pi(k+1/2)/N) + gk,i sin (pi(k+1/2)/N); where T1 and T2 are the subtraction stream and addition stream coefficients, respectively. However, Fielder discloses, as stated above (see claim 1), a starting equitation (24) with condition equations (6 and 25) that corresponds to eq. 1 of the specification, intermediate equations (26, 27) that corresponds to intermediate eq. 13 of the specification, and final result equation (28) that corresponds to eq. 16 of the specification, in which the claimed limitation can be proven or derived from the referenced prior art equations through a mathematical reasoning, which also means that the equation 28 can be reasoned to derive the claimed terms. Further, according to eq. (28) (column 36) and eq. (6) (column 18), the angle in

eq. (28) can be separated into to two angles: 2pi (k+ ½) m/N = 2pi (k+ ½)/4 + pi (k+ ½), where m=(N/2 +1)/2, so that the trigonometric function factor of sum of two angles can be expressed by a sum (two term) of trigonometric function factors of individual angles, such as cos(a+b) = cos (a) cos(b) + sin(a)sin(b), thus one of the trigonometric function factors can read on the claimed limitation. Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to compute Fielder's equations by mathematical computing or reasoning, including intermediate steps and result as claimed, for the purpose of providing a complete computation algorithm from a staring point (equation 24) to reach the final conclusion (equation 28) (Fielder: column 35, lines 60-67).

As per **claim 9** (depending on claim 1), Fielder does not expressly disclose that steps iv) and v) comprise generating the audio coded frequency domain coefficients Xk according to: Xk = T1 cos (pi(2k+1)/4) - T2 sin (pi(2k+1)/4); where Xk are the audio coded frequency domain coefficients; and cos (pi(2k+1)/4) and sin (pi(2k+1)/4 are the third trigonometric function factors. However, Fielder discloses, as stated above (see claim 1), a starting equitation (24) with condition equations (6 and 25) that corresponds to eq.1 of the specification, intermediate equations (26, 27) that corresponds to intermediate eq. 13 of the specification, and final result equation (28) (see column 35, line 33 to column 36, line 35) that corresponds to eq. 16 of the specification, in which the claimed limitation can be proven or derived from the referenced prior art equations through a mathematical reasoning, which suggests that the equation can be reasoned to derive the claimed terms. Further, according to eq. (28) (column 36) and eq. (6) (column 18), the angle in the eq. (28) can be separated into to two angles: 2pi (k+ ½) m/N = 2pi (k+ ½)/4 + pi (k+ ½)/N, where m=(N/2 +1)/2, so that the trigonometric function factor with a

sum of two angles can be expressed by a sum (two term) of trigonometric function factors with individual angles, such as cos(a+b) = cos (a) cos(b) + sin(a)sin(b), thus, one of the trigonometric function factors can read on the claimed limitation. Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to compute Fielder' the equations by a mathematical reasoning, including intermediate steps and result as claimed, for the purpose of providing a complete computation algorithm from a staring pcint (equation 24) to reach the final conclusion (equation 28) (Fielder: column 35, lines 60-67).

As per **claim 17**, the rejection is based on the same reason described for claim 1, because claim 17 recites same or similar limitation(s) as claim 1.

As per **claim 18** (depending on claim 17), Fielder further discloses that the pre-multiplication factor, and first and second post-multiplication factors are trigonometric function factors (column 36, equations (26) and (28), wherein factor exp(-j πn/N) = cos(- πn/N) + j sin (-πn/N), and term of cos [2pi (k+ ½) m/N] = cos [2pi (k+ ½)/4 + pi (k+ ½)] when using equation 16: m=(N/2 +1)/2; so that by further using a trigonometric property of cos(a+b) = cos (a) cos(b) - sin(a)sin(b), the term of cos() or sin() in equation 28 produces two trigonometric function factors, which reads on the claimed limitation).

As per **claims 19-21** (depending on claim 17), the rejection is based on the same reason described for claims 3-5 respectively, because claims 19-21 recites same or similar limitation(s) as claims 3-5 respectively.

As per **claim 22** (depending on claim 17), Fielder further discloses that the pre-processing operations are performed on each sample in the input sequence individually (column

36, equation 27, which shows that the operation is performed on each sample in input x(n) individually).

As per **claim 23** (depending on claim 17), Fielder further discloses that the post-processing operations are performed on each transform coefficient in the sequence individually, (column 36, equation 28, which shows that the post-processing operation is performed on each transform coefficient R(k) and Q(k) individually).

9. Claims 10-13, 16 and 24-27 are rejected under 35 U.S.C. 103(a) as being unpatentable over Fielder in view of Proakis et al. ("Digital Signal Processing, principles, algorithms, and applications", 3rd Edition, 1996, ISBN 0-13-373762-4) hereinafter referenced as Proakis.

As per **claim 10**, Fielder discloses a method and apparatus for encoding and decoding audio information, comprising:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence (column 16, line 40 to column 17, line 11 'a single FFT can be used to perform the DCT and DST simultaneously by defining them respectively as the real and imaginary components of a signal complex transform' and 'processing a signal sample block from each of the two channels', which suggest that the signal uses the real component for one channel and imaginary components for another channel);

determining a Fourier transform coefficient sequence (column 16, lines 40-55, 'a single FFT can be used to perform the DCT and DST simultaneously by define them respectively as the real and imaginary components of a signal complex transform'; column 36, lines 9-35 and

equations 27 and 28, where in the signal x(n) has real and imaginary components: x(n)=xr(n)+jxi(n));

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels (column 16, lines 40-55, 'a single FFT can be used to perform the DCT and DST simultaneously by define them respectively as the real and imaginary components of a signal complex transform'; column 36, lines 20-55 and equation 28, 'In two-channel systems, signal sample blocks from each of two channels are transformed by FFT processes into DCT1/DCT2 block pair').

Even though, as stated above, Fielder discloses that a single FFT can be used to perform the DCT and DST simultaneously by defining them respectively as the real and imaginary components of a single complex transform (column 16, lines 40-55), and further discloses some the intermediate results or steps of processing transform coefficient sequences (equations 6, 24, 26, 27 and 28 and column 35, line 32 to column 36, lines 67), Fielder does not expressly disclose all detailed intermediate steps for "generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence". However, this feature is well known in the art as evidenced by Proakis, who teaches symmetry properties of the discrete-time Fourier transform (page 290-291) that disclose the mathematical relationships between different time domain/frequency domain signal components, including even/odd, real/image, and conjugate relations (equations 4.3.37 and 5.2.31, Tables 4.4 and 5.1, and Fig. 4.29), specially combining the third and fourth properties in Tables 4.4 and 5.1, which corresponds the claimed limitation.

Fielder further discloses an efficient computation of the DFT of two real sequences (page 475-476) that can compute two real signal sequences in a complex-valued sequence by performing a single DFT (FFT), so that the respective sequences of audio frequency domain coefficient sequences for the two real signal sequences (corresponding to two audio channel signals) can be derived by using the FFT transformed coefficients and the symmetry properties. Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder by specifically providing a FFT algorithm to perform a single DFT for two real signal (two channel) sequences by using the symmetry properties of the Fourier transform, as taught by Proakis, for the purpose of enhancing the efficiency of the FFT algorithm (Proakis: page 475, paragraph 6).

As per **claim 11** (depending on claim 10), Fielder in view of Proakis further discloses that for each corresponding coefficient in the first and second transform coefficient sequences, selecting first and second transform coefficients from said Fourier transform coefficient sequence, determining a complex conjugate of said second transform coefficient, combining said first transform coefficient and said complex conjugate for said first transform coefficient sequence and differencing said first transform coefficient and said complex conjugate for said second transform coefficient sequence, (Fielder: column 36, lines 35 and equations 27, 28 and 6; Proakis: pages 290-291, equation 4.3.37 and Table 4.4, wherein two time domain signal sequences can be defined as a complex sequence: x(n) = xr(n) + jxi(n); the frequency domain sequence can be expressed by: X(k)=FFT[xr(n)exp(-jnpi/N)+jxi(n)exp(-jnpi/N)] = Xr(k)+jXi(k), which corresponds to equation 27 of Fielder; and the frequency domain sequence can be further expressed by: X(k)= Xr(k)+jXi(k) = [Xre(k)+jXio(k)] + [Xro(k)+jXie(k)] = Xxr(k) +Xxi(k),

wherein the subscripts indicate: r -- real part, i –imaginary part, e – even part, o – odd part, x – corresponding input signal sequences, which corresponds to term R(k) and Q(k) in equation 28 of Fielder by combining symmetry properties on equation 4.4.37 (Proakis: page 290) and complex conjugate process in Table 5.1 (Proakis: page 415), wherein equation 28 of Fielder has same form but the R(k) and Q(k) include both components from the first and second signals xr and xi, and using third and fourth properties in Table 4.4 or 5.1, two audio signal frequency sequences can be obtained).

As per **claim 12** (depending on claim 10), the rejection is based on the same reason described for claim 6, because claim 12 recites same or similar limitation(s) as claim 6.

As per **claim 13** (depending on claim 11), Fielder in view of Proakis further discloses a properties of DTFT: Xe(k)=1/2[X(k)+X*(N-k)] and Xo(k)=1/2[X(k)-X*(N-k)] (Proakis: page 415, Table 5.1) and the derived equations for computation of the DFT of two real sequences (Proakis: page 476, equations 6.2.7 and 6.2.8), where e indicates even part, o indicates odd part, and X(k) corresponds to coefficient X*(k) in equation 27 of Fielder (Fielder: column 36, lines 1-35), so that the combined teachings correspond to the clamed "said first and second transform coefficient sequences are generated according to: Gk (Zk + Z*N-k-1)/2, G'k (Zk - Z*N-k-1)/2j where Gk is said first transform coefficient sequence; G'k is said second transform coefficient sequence; N is the number of input sequence audio samples; k = 0,..., (N/2 - 1); Zk is said first transform coefficient; Z*N-k-1 is the complex conjugate of said second transform coefficient; and j is the complex constant".

As per **claim 16** (depending on claim 10), Fielder in view of Proakis further discloses

applying a windowing function in combination with multiplying the complex sample sequence by a first trigonometric function factor (Fig. 1a, 'analysis widow'; Figs. 6a-6d).

As per **claim 24**, it recites audio coding method, which corresponds to the combination of claims 1, 10 and 13. The rejection is based on the same reason described for claims 1,10 and 13, because claim 24 recites same or similar limitation(s) as claims 1,10 and 13.

As per **claims 25** (depending on claim 24), the rejection is based on the same reason described for claim 3, because claim 25 recites same or similar limitation(s) as claim 3.

As per **claims 26** (depending on claim 24), the rejection is based on the same reason described for claim 18, because claim 26 recites same or similar limitation(s) as claim 18.

As per **claim 27**, Fielder discloses a method and apparatus for encoding and decoding audio information, comprising:

obtaining first and second input sequences of digital audio samples x[n], y[n] corresponding to respective first and second audio channels, (column 16, line 40 to column 17, line 11, 'both input signal sample blocks consist only real-valued samples' and 'processing a signal sample block from each of the two channels');

combining the first and second input sequences of digital audio samples into a single complex input sample sequence z[n], where z[n] = x[n] + jy[n], (column 16, lines 43-64; 'a single FFT can be used to perform the DCT and DST simultaneously by define them respectively as the real and imaginary components of a signal complex transform', which suggest that the input signals includes a real component for one channel and an imaginary components for another channel);

pre-processing the complex input sequence samples including applying a pre-multiplication factor cos(pi n/N) +jsin(pi n/N) to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and n = 0,..., (N-1) (column 36, equations (26) and (27), wherein a factor of exp(-j πn/N) = cos(- πn/N) + j sin (- πn/N) is used for pre-multiplying, which has equivalent functionality of the claimed limitation because one of intermediate results can be derived from the other by conjugate processing (based on eqs. 12 and 13 of the specification));

transforming the modified complex input sequence samples into a complex transform coefficient sequence Zk utilizing a fast Fourier transform, wherein k = 0,..., (N/2- 1), (column 16, lines 40-55, 'a single FFT can be used to perform the DCT and DST simultaneously by define them respectively as the real and imaginary components of a signal complex transform'; column 36, lines 9-35 and equations 27, 'signal sample blocks from the two channels are transformed by FFT processes into DCT1/DCT2 block pair', wherein input sequence x(n) may have two sequences combined in a complex sequence, so that in this case it is obvious that equation 27 would be: X*(k)= FFT[xr(n)exp(-jnpi/N) + jxi(n)exp(-jnpi/N)]; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients (column 36, lines 9-35 and equations 28).

But, Fielder does not expressly disclose "to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels Xk, Yk" according to claimed intermediate steps of computing the process for a two-channel system using complex signal in an FFT transform. However, this feature is well known in the art as

evidenced by Proakis, who teaches symmetry properties of the discrete-time Fourier transform (page 290-291) that disclose the mathematical relationships between different time domain/frequency domain signal components, and efficient computation of the DFT of two real sequences (page 475-476) that combines the two real signal sequences, such as two channel signal sequences, into a complex-valued sequence for performing a single DFT (FFT), so that the respective sequences of audio frequency domain coefficient sequences for the two real signal sequences (corresponding to two audio channel signals) can be derived by using the FFT transformed coefficients and the symmetry properties. Particularly, Proakis discloses equations 6.2.7 and 6.2.8 (page 476) that are equivalent to the claimed Gk and G'k, and symmetry equation 5.2.31 (page 415), which can be used in equation 28 of Fielder to generate the claimed result by mathematically reasoning: let a=2pi(k+1/2)/4, b=pi(k+1/2)/N, Xr=R(k), Xi= -Q(k), X(k)=(X*(k))*=[R(k)-jQ(k)]= Xr+jXi (Fielder: eq. 27); and take m=(1+N/2)/2 (Fielder: eq.16), then from eq.28 (Fielder):

C(k)= R(k)cos (a+b) + Q(k)sin(a+b) = Xr cos (a+b) - Xi sin(a+b)

= Xr [cos (a)cos (b) - (sin(a)sin(b)] - Xi[sin(a)cos(b)+cos(a)sin(b)]

=cos(b)[Xr cos(a) - Xi sin(a)] – sin(b)[Xr sin (a) + Xi cos(a)]

= cos(b)[(Xre+Xro)cos(a)–(Xie+Xio)sin(a)] – sin(b)[(Xre+Xro)sin(a)+(Xie+Xio)cos(a)]

where, subscripts indicate: r—real part, i—imaginary part, e—even part, o—odd part, since Proakis teaches that even part of frequency coefficient: X1(k) = [X(k)+(X*(N-k)]/2=Xre+jXie corresponds to real part of input sequence x1(n) and odd part of frequency coefficient X2(k)=[X(k)+(X*(N-k)]/j2=Xro+jXio corresponds to imaginary part of input sequence x2(n) (Proakis: page 415, Table 5.1 and equation 5.2.31; page 476, equations 6.2.7 and 6.2.8), thus,

separating the even and odd parts into two groups would lead to obtain the respective frequency coefficients corresponding to the two input signal sequences as claimed.

Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder by specifically providing a FFT algorithm by performing a single DFT for two real signal (two channel) sequences by using the symmetry properties of the Fourier transform, as taught by Proakis, for the purpose of enhancing the efficiency of the FFT algorithm (Proakis: page 475, paragraph 6).

10. Claims 14-15 and 28-39 are rejected under 35 U.S.C. 103(a) as being unpatentable over Fielder in view of Proakis and further in view of Jhung (US 6304847 B1).

As per **claim 14** (depending on claim 10), even though Fielder teaches the tradeoff of using longer or shorter block length for a transform (column 3, lines 30-67), Fielder in view of Proakis does not expressly disclose "examining said first and second sequences of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined". However, this feature is well known in the art as evidenced by Jhung, who discloses that the Dolly AC-3 standard utilizes long transform or two short transform based on the transition condition (column 3, line 62 to column 4, line 24). Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder in view of Proakis by specifically providing long transform or two short transform based on the transition condition, as taught by Jhung, for the purpose of handling different transition situations (Proakis: column 3, line 63 to column 4, line 2).

As per **claim 15** (depending on claim 10), Fielder teaches the tradeoff of using longer or shorter block length for a transform (column 3, lines 30-67) and "pairing the channels according to their determined transform length, and coding the audio samples of first and second channels in each pair according to determined transform length", (column 17, lines 3-25, 'two-channel system', processing a signal sample block (necessarily including a determined transform length) from each of the two channels: a DCT block...and A DST block', 'the coded (coding) block for given channel alternate (pairing) between the DCT and DST', 'a pair of blocks, one for each channel, are quantized and formatted (coding)'). But, Fielder in view of Proakis does not expressly disclose "determining a transform length for each of the channels". However, this feature is well known in the art as evidenced by Jhung, who discloses that the Dolly AC-3 standard utilizes long transform or two short transform based on the transition condition (determining transform length) (column 3, line 62 to column 4, line 24). Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder in view of Proakis by specifically providing a long transform or two short transform based on the transition condition (determining transform length) as taught by Jhung, for the purpose of handling different transition situations (Proakis: column 3, line 63 to column 4, line 2).

As per **claims 28-39**, they recite an apparatus for coding input audio samples. The rejection is based on the same reason described for claims 1-2, 18, 3-5, 22-23, 14, 10 and 38-39 respectively, because claims 28-39 recite same or similar limitation(s) as claims 1-2, 18, 3-5, 22-23, 14, 10 and 38-39 respectively.

Conclusion

11. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL.** See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a). A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

12. Any response to this action should be mailed to:

Mail Stop AF
Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450

or faxed to:

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Or:

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Patent Correspondence delivered by hand or delivery services, other than the USPS, should be addressed as follows and brought to U.S. Patent and Trademark Office, 220 20th Street S., Customer Window, **Mail Stop AF**, Crystal Plaza Two, Lobby, Room 1B03, Arlington, VA, 22202

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Qi Han whose telephone numbers is (703) 305-5631. The examiner can normally be reached on Monday through Thursday from 9:00 a.m. to 7:00 p.m. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil, can be reached on (703) 305-6954.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Inquiries regarding the status of submissions relating to an application or questions on the Private PAIR system should be directed to the Electronic Business Center (EBC) at 866-217-9197 (toll-free) or 703-305-3028 between the hours of 6 a.m. and midnight Monday through Friday EST, or by e-mail at: ebc@uspto.gov. For general information about the PAIR system, see http://pair-direct.uspto.gov.

QH/qh
December 3, 2004

RICHEMOND DORVIL
SUPERVISORY PATENT EXAMINER

OIPE JCW
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PATENT & TRADEMARK OFFICE

FORM PTO-1449 (REV.7-80)	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTY. DOCKET NO. 851663.413USPC	APPLICATION NO. 09/622,736
INFORMATION DISCLOSURE STATEMENT (Use several sheets if necessary)		APPLICANTS Mohammed Javed Absar et al.	
		FILING DATE October 27, 2000	GROUP ART UNIT 2654

U.S. PATENT DOCUMENTS

*EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
	AA	5,181,183	01/19/93	Miyazaki	364	725	
JH	AB	5,592,584	01/07/97	Ferreira et al.	395	2.12	
	AC						
	AD						
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FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	TRANSLATION YES	TRANSLATION NO
	AK	0 590 790 A2	04/06/94	EP		
JH	AL	0 718 746 A1	06/26/96	EP (+ Abstract in English)		
	AM	0 564 089 B1	01/20/99	EP		
	AN	0 506 111 B1	04/12/00	EP		
	AO					

OTHER PRIOR ART *(Including Author, Title, Date, Pertinent Pages, Etc.)*

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EXAMINER	DATE CONSIDERED 12/2/04

* EXAMINER: Initial if reference considered, whether or not criteria is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant(s).

Office Action Summary	Application No.	Applicant(s)	
	09/622,736	ABSAR ET AL.	
	Examiner	Art Unit	
	Qi Han	2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 27 August 2004.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-39 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-39 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1)☐ Notice of References Cited (PTO-892)

2)☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3)☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 08/27/2004.

4)☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5)☐ Notice of Informal Patent Application (PTO-152)

6)☐ Other: _____.

APPENDIX C

PROAKIS, ET AL., DIGITAL SIGNAL PROCESSING, PRINCIPLES, ALGORITHMS AND APPLICATIONS (3D ED. 1996)

CITED BY EXAMINER IN OFFICE ACTION MAILED APRIL 23, 2004

V

Digital Signal Processing

Principles, Algorithms, and Applications

Third Edition

John G. Proakis
Northeastern University

Dimitris G. Manolakis
Boston College

PRENTICE HALL
Upper Saddle River, New Jersey 07458

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Similarly, if $x_I(n)$ is even [i.e., $x_I(-n) = x_I(n)$], we have

$$X_R(\omega) = 0 \tag{4.3.33}$$

$$X_I(\omega) = x_I(0) + 2\sum_{n=1}^{\infty} x_I(n)\cos\omega n \qquad \text{(even)} \tag{4.3.34}$$

$$x_I(n) = \frac{1}{\pi}\int_0^{\pi} X_I(\omega)\cos\omega n \, d\omega \tag{4.3.35}$$

An arbitrary, possibly complex-valued signal $x(n)$ can be decomposed as

$$\begin{aligned} x(n) = x_R(n) + jx_I(n) &= x_R^e(n) + x_R^o(n) + j[x_I^e(n) + x_I^o(n)] \\ &= x_e(n) + x_o(n) \end{aligned} \tag{4.3.36}$$

where, by definition,

$$x_e(n) = x_R^e(n) + jx_I^e(n) = \tfrac{1}{2}[x(n) + x^*(-n)]$$

$$x_o(n) = x_R^o(n) + jx_I^o(n) = \tfrac{1}{2}[x(n) - x^*(-n)]$$

The superscripts e and o denote the even and odd signal components, respectively. We note that $x_e(n) = x_e(-n)$ and $x_o(-n) = -x_o(n)$. From (4.3.36) and the Fourier transform properties established above, we obtain the following relationships:

$$\begin{array}{c} x(n) = [x_R^e(n) + jx_I^e(n)] + [x_R^o(n) + jx_I^o(n)] \\ \updownarrow \qquad\quad \updownarrow \qquad\qquad \times \\ X(\omega) = [X_R^e(\omega) + jX_I^e(\omega)] + [X_R^o(\omega) + jX_I^o(\omega)] \end{array} \tag{4.3.37}$$

These symmetry properties of the Fourier transform are summarized in Table 4.4 and in Fig. 4.29. They are often used to simplify Fourier transform calculations in practice.

Example 4.3.1

Determine and sketch $X_R(\omega)$, $X_I(\omega)$, $|X(\omega)|$, and $\measuredangle X(\omega)$ for the Fourier transform

$$X(\omega) = \frac{1}{1 - ae^{-j\omega}} \qquad -1 < a < 1 \tag{4.3.38}$$

Solution By multiplying both the numerator and denominator of (4.3.38) by the complex conjugate of the denominator, we obtain

$$X(\omega) = \frac{1 - ae^{j\omega}}{(1 - ae^{-j\omega})(1 - ae^{j\omega})} = \frac{1 - a\cos\omega - ja\sin\omega}{1 - 2a\cos\omega + a^2}$$

This expression can be subdivided into real and imaginary parts. Thus we obtain

$$X_R(\omega) = \frac{1 - a\cos\omega}{1 - 2a\cos\omega + a^2}$$

$$X_I(\omega) = -\frac{a\sin\omega}{1 - 2a\cos\omega + a^2}$$

Substitution of the last two equations into (4.3.15) and (4.3.16) yields the magnitude and phase spectra as

$$|X(\omega)| = \frac{1}{\sqrt{1 - 2a\cos\omega + a^2}} \tag{4.3.39}$$

TABLE 4.4 SYMMETRY PROPERTIES OF THE DISCRETE-TIME FOURIER TRANSFORM

Sequence	DTFT
$x(n)$	$X(\omega)$
$x^*(n)$	$X^*(-\omega)$
$x^*(-n)$	$X^*(\omega)$
$x_R(n)$	$X_e(\omega) = \frac{1}{2}[X(\omega) + X^*(-\omega)]$
$jx_I(n)$	$X_o(\omega) = \frac{1}{2}[X(\omega) - X^*(-\omega)]$
$x_e(n) = \frac{1}{2}[x(n) + x^*(-n)]$	$X_R(\omega)$
$x_o(n) = \frac{1}{2}[x(n) - x^*(-n)]$	$jX_I(\omega)$
Real Signals	
Any real signal $x(n)$	$X(\omega) = X^*(-\omega)$ $X_R(\omega) = X_R(-\omega)$ $X_I(\omega) = -X_I(-\omega)$ $\lvert X(\omega)\rvert = \lvert X(-\omega)\rvert$ $\measuredangle X(\omega) = -\measuredangle X(-\omega)$
$x_e(n) = \frac{1}{2}[x(n) + x(-n)]$ (real and even)	$X_R(\omega)$ (real and even)
$x_o(n) = \frac{1}{2}[x(n) - x(-n)]$ (real and odd)	$jX_I(\omega)$ (imaginary and odd)

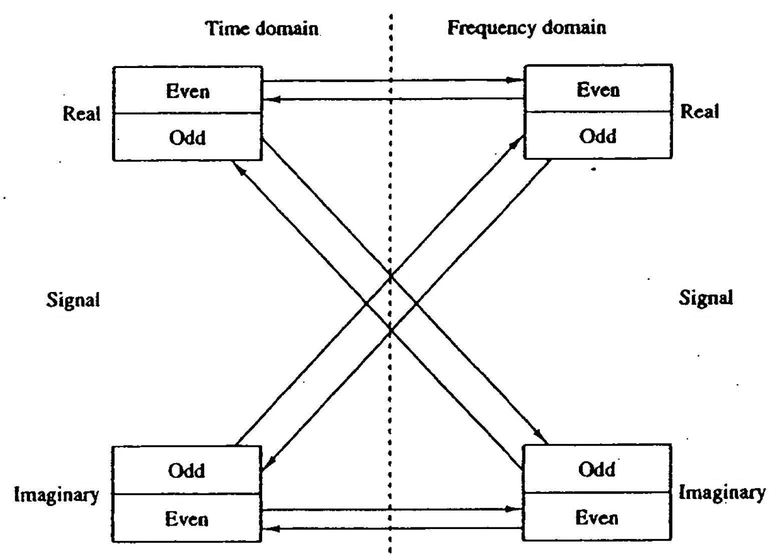

Figure 4.29 Summary of symmetry properties for the Fourier transform.

TABLE 5.1 SYMMETRY PROPERTIES OF THE DFT

N-Point Sequence $x(n)$, $0 \le n \le N-1$	N-Point DFT
$x(n)$	$X(k)$
$x^*(n)$	$X^*(N-k)$
$x^*(N-n)$	$X^*(k)$
$x_R(n)$	$X_{ce}(k) = \frac{1}{2}[X(k) + X^*(N-k)]$
$jX_I(n)$	$X_{co}(k) = \frac{1}{2}[X(k) - X^*(N-k)]$
$x_{ce}(n) = \frac{1}{2}[x(n) + x^*(N-n)]$	$X_R(k)$
$x_{co}(n) = \frac{1}{2}[x(n) - x^*(N-n)]$	$jX_I(k)$
Real Signals	
Any real signal $x(n)$	$X(k) = X^*(N-k)$ $X_R(k) = X_R(N-k)$ $X_I(k) = -X_I(N-k)$ $\lvert X(k)\rvert = \lvert X(N-k)\rvert$ $\angle X(k) = -\angle X(N-k)$
$x_{ce}(n) = \frac{1}{2}[x(n) + x(N-n)]$	$X_R(k)$
$x_{co}(n) = \frac{1}{2}[x(n) - x(N-n)]$	$jX_I(k)$

The symmetry properties given above may be summarized as follows:

$$\begin{array}{c} x(n) = x_R^e(n) + x_R^o(n) + jx_I^e(n) + jx_I^o(n) \\ \updownarrow \\ X(k) = X_R^e(k) + X_R^o(k) + jX_I^e(k) + jX_I^o(k) \end{array} \tag{5.2.31}$$

All the symmetry properties of the DFT can easily be deduced from (5.2.31). For example, the DFT of the sequence

$$x_{pe}(n) = \tfrac{1}{2}[x_p(n) + x_p^*(N-n)]$$

is

$$X_R(k) = X_R^e(k) + X_R^o(k)$$

The symmetry properties of the DFT are summarized in Table 5.1. Exploitation of these properties for the efficient computation of the DFT of special sequences is considered in some of the problems at the end of the chapter.

5.2.2 Multiplication of Two DFTs and Circular Convolution

Suppose that we have two finite-duration sequences of length N, $x_1(n)$ and $x_2(n)$. Their respective N-point DFTs are

$$X_1(k) = \sum_{n=0}^{N-1} x_1(n)e^{-j2\pi nk/N} \qquad k = 0, 1, \ldots, N-1 \tag{5.2.32}$$

$$X_2(k) = \sum_{n=0}^{N-1} x_2(n)e^{-j2\pi nk/N} \qquad k = 0, 1, \ldots, N-1 \tag{5.2.33}$$

Finally, we note that the emphasis in our discussion of FFT algorithms was on radix-2, radix-4, and split-radix algorithms. These are by far the most widely used in practice. When the number of data points is not a power of 2 or 4, it is a simple matter to pad the sequence $x(n)$ with zeros such that $N = 2^\nu$ or $N = 4^\nu$.

The measure of complexity for FFT algorithms that we have emphasized is the required number of arithmetic operations (multiplications and additions). Although this is a very important benchmark for computational complexity, there are other issues to be considered in practical implementation of FFT algorithms. These include the architecture of the processor, the available instruction set, the data structures for storing twiddle factors, and other considerations.

For general-purpose computers, where the cost of the numerical operations dominate, radix-2, radix-4, and split-radix FFT algorithms are good candidates. However, in the case of special-purpose digital signal processors, featuring single-cycle multiply-and-accumulate operation, bit-reversed addressing, and a high degree of instruction parallelism, the structural regularity of the algorithm is equally important as arithmetic complexity. Hence for DSP processors, radix-2 or radix-4 decimation-in-frequency FFT algorithms are preferable in terms of speed and accuracy. The irregular structure of the SRFFT may render it less suitable for implementation on digital signal processors. Structural regularity is also important in the implementation of FFT algorithms on vector processors, multiprocessors, and in VLSI. Interprocessor communication is an important consideration in such implementations on parallel processors.

In conclusion, we have presented several important considerations in the implementation of FFT algorithms. Advances in digital signal processing technology, in hardware and software, will continue to influence the choice among FFT algorithms for various practical applications.

6.2 APPLICATIONS OF FFT ALGORITHMS

The FFT algorithms described in the preceding section find application in a variety of areas, including linear filtering, correlation, and spectrum analysis. Basically, the FFT algorithm is used as an efficient means to compute the DFT and the IDFT.

In this section we consider the use of the FFT algorithm in linear filtering and in the computation of the crosscorrelation of two sequences. The use of the FFT in spectrum analysis is considered in Chapter 12. In addition we illustrate how to enhance the efficiency of the FFT algorithm by forming complex-valued sequences from real-valued sequences prior to the computation of the DFT.

6.2.1 Efficient Computation of the DFT of Two Real Sequences

The FFT algorithm is designed to perform complex multiplications and additions, even though the input data may be real valued. The basic reason for this situation is

that the phase factors are complex and hence, after the first stage of the algorithm, all variables are basically complex-valued.

In view of the fact that the algorithm can handle complex-valued input sequences, we can exploit this capability in the computation of the DFT of two real-valued sequences.

Suppose that $x_1(n)$ and $x_2(n)$ are two real-valued sequences of length N, and let $x(n)$ be a complex-valued sequence defined as

$$x(n) = x_1(n) + jx_2(n) \qquad 0 \le n \le N-1 \tag{6.2.1}$$

The DFT operation is linear and hence the DFT of $x(n)$ can be expressed as

$$X(k) = X_1(k) + jX_2(k) \tag{6.2.2}$$

The sequences $x_1(n)$ and $x_2(n)$ can be expressed in terms of $x(n)$ as follows:

$$x_1(n) = \frac{x(n) + x^*(n)}{2} \tag{6.2.3}$$

$$x_2(n) = \frac{x(n) - x^*(n)}{2j} \tag{6.2.4}$$

Hence the DFTs of $x_1(n)$ and $x_2(n)$ are

$$X_1(k) = \frac{1}{2}\{DFT[x(n)] + DFT[x^*(n)]\} \tag{6.2.5}$$

$$X_2(k) = \frac{1}{2j}\{DFT[x(n)] - DFT[x^*(n)]\} \tag{6.2.6}$$

Recall that the DFT of $x^*(n)$ is $X^*(N-k)$. Therefore,

$$X_1(k) = \frac{1}{2}[X(k) + X^*(N-k)] \tag{6.2.7}$$

$$X_2(k) = \frac{1}{j2}[X(k) - X^*(N-k)] \tag{6.2.8}$$

Thus, by performing a single DFT on the complex-valued sequence $x(n)$, we have obtained the DFT of the two real sequences with only a small amount of additional computation that is involved in computing $X_1(k)$ and $X_2(k)$ from $X(k)$ by use of (6.2.7) and (6.2.8).

6.2.2 Efficient Computation of the DFT of a 2*N*-Point Real Sequence

Suppose that $g(n)$ is a real-valued sequence of $2N$ points. We now demonstrate how to obtain the $2N$-point DFT of $g(n)$ from computation of one N-point DFT involving complex-valued data. First, we define

$$\begin{aligned} x_1(n) &= g(2n) \\ x_2(n) &= g(2n+1) \end{aligned} \tag{6.2.9}$$

Thus we have subdivided the $2N$-point real sequence into two N-point real sequences. Now we can apply the method described in the preceding section.

Let $x(n)$ be the N-point complex-valued sequence

$$x(n) = x_1(n) + jx_2(n) \tag{6.2.10}$$

From the results of the preceding section, we have

$$\begin{aligned} X_1(k) &= \frac{1}{2}[X(k) + X^*(N-k)] \\ X_2(k) &= \frac{1}{2j}[X(k) - X^*(N-k)] \end{aligned} \tag{6.2.11}$$

Finally, we must express the $2N$-point DFT in terms of the two N-point DFTs, $X_1(k)$ and $X_2(k)$. To accomplish this, we proceed as in the decimation-in-time FFT algorithm, namely,

$$\begin{aligned} G(k) &= \sum_{n=0}^{N-1} g(2n)W_{2N}^{2nk} + \sum_{n=0}^{N-1} g(2n+1)W_{2N}^{(2n+1)k} \\ &= \sum_{n=0}^{N-1} x_1(n)W_N^{nk} + W_{2N}^k \sum_{n=0}^{N-1} x_2(n)W_N^{nk} \end{aligned}$$

Consequently,

$$\begin{aligned} G(k) &= X_1(k) + W_2^k N X_2(k) \qquad k = 0, 1, \ldots, N-1 \\ G(k+N) &= X_1(k) - W_2^k N X_2(k) \qquad k = 0, 1, \ldots, N-1 \end{aligned} \tag{6.2.12}$$

Thus we have computed the DFT of a $2N$-point real sequence from one N-point DFT and some additional computation as indicated by (6.2.11) and (6.2.12).

6.2.3 Use of the FFT Algorithm in Linear Filtering and Correlation

An important application of the FFT algorithm is in FIR linear filtering of long data sequences. In Chapter 5 we described two methods, the overlap-add and the overlap-save methods for filtering a long data sequence with an FIR filter, based on the use of the DFT. In this section we consider the use of these two methods in conjunction with the FFT algorithm for computing the DFT and the IDFT.

Let $h(n)$, $0 \le n \le M-1$, be the unit sample response of the FIR filter and let $x(n)$ denote the input data sequence. The block size of the FFT algorithm is N, where $N = L + M - 1$ and L is the number of new data samples being processed by the filter. We assume that for any given value of M, the number L of data samples is selected so that N is a power of 2. For purposes of this discussion, we consider only radix-2 FFT algorithms.

The N-point DFT of $h(n)$, which is padded by $L-1$ zeros, is denoted as $H(k)$. This computation is performed once via the FFT and the resulting N complex numbers are stored. To be specific we assume that the decimation-in-frequency

APPENDIX D

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(71) Applicant *(for all designated States except US)*: SGS–THOMSON MICROELECTRONICS ASIA PACIFIC (PTE) LTD [SG/SG]; 28 Ang Mo Kio Industrial Park 2, Singapore 569508 (SG).

(72) Inventors; and
(75) Inventors/Applicants *(for US only)*: ABSAR, Mohammed, Javed [IN/SG]; Block 411, #09–1006 Hougang Avenue 10, Singapore 530411 (SG). GEORGE, Sapna [IN/SG]; Block 315, #06–220 Serangoon Avenue 2, Singapore 550315 (SG). ALVAREZ–TINOCO, Antonio, Mario [GB/SG]; 32 Toh Tuck Road #03–01, Singapore 596710 (SG).

(74) Agent: DONALDSON & BURKINSHAW; P.O. Box 3667, Singapore 905667 (SG).

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(54) Title: A FAST FREQUENCY TRANSFORMATION TECHIQUE FOR TRANSFORM AUDIO CODERS

(57) Abstract

A method for coding digital audio data in which coded Fast Modified Discrete Cosine Transform (FMDCT) coefficients are computed utilising a Fast Fourier Transform (FFT) method. The described method allows a significant reduction in computations as compared to an ordinary DCT coding procedure. Also, pairs of audio channels can be combined to use a single FFT computation, where the selected transform length for the paired channels is the same. In such cases where pairing of identical transform length channels is not possible, a long transform length channel is combined with a short transform length channel and converted in two short transforms. A windowing function is also combined with a pre–processing stage to the transformation, further decreasing computational requireements.

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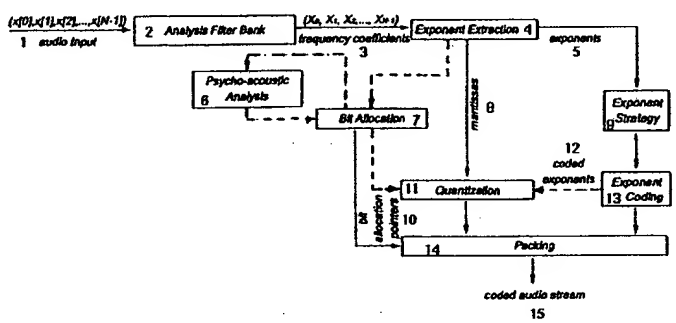

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A FAST FREQUENCY TRANSFORMATION TECHNIQUE FOR TRANSFORM AUDIO CODERS

Technical Field

This invention is applicable in the field of multi-channel audio coders which use modified discrete cosine transform as a step in the compression of audio signals.

Background Art

In order to more efficiently broadcast or record audio signals, the amount of information required to represent the audio signals may be reduced. In the case of digital audio signals, the amount of digital information needed to accurately reproduce the original pulse code modulation (PCM) samples may be reduced by applying a digital compression algorithm, resulting in a digitally compressed representation of the original signal. The goal of the digital compression algorithm is to produce a digital representation of an audio signal which, when decoded and reproduced, sounds the same as the original signal, while using a minimum of digital information for the compressed or encoded representation.

Recent advances in audio coding technology have led to high compression ratios while keeping audible degradation in the compressed signal to a minimum. These coders are intended for a variety of applications, including 5.1 channel film soundtracks, HDTV, laser discs and multimedia. Description of one applicable method can be found in the Advanced Television Systems Committee (ATSC) Standard document entitled "Digital Audio Compression (AC-3) Standard", Document A/52, 20 December, 1995.

In the basic approach, at the encoder the time domain audio signal is first converted to the frequency domain using a bank of filters. The frequency domain coefficients, thus generated, are converted to fixed point representation. In fixed point syntax, each coefficient is represented as a mantissa and an exponent. The bulk of the compressed bitstream transmitted to the decoder comprises these exponents and mantissas.

The exponents are usually transmitted in their original form. However, each mantissa must be truncated to a fixed or variable number of decimal places. The number of bits to be used for coding each mantissa is obtained from a bit allocation algorithm which may be based on the masking property of the human auditory system. Lower numbers of bits result in higher compression ratios because less space is required to transmit the coefficients. However, this may cause high quantization errors, leading to audible distortion. A good distribution of available bits to each mantissa forms the core of the advanced audio coders.

The frequency transformation phase has one of the greatest computation requirements in a transform coder. Therefore, an efficient implementation of this phase can decrease the computation requirement of the system significantly and make real time operation of the encoder more easily attainable.

In some encoders such as those specified in the AC-3 standard, the frequency domain transformation of signals is performed by the modified discrete cosine transform (MDCT). If directly implemented, the MDCT requires $O(N^2)$ additions and multiplications. However it has been found possible to reduce the number of required operations significantly if the MDCT equation is able to be computed in a from that is amenable to the use of the well known Fast Fourier Transform (FFT) method of J.W. Cooley and J.W. Tukey (1960). Moreover, using a single FFT for two channels can result in greater reduction in computational requirements of the system.

<u>Summary of the Invention</u>

In accordance with the present invention there is provided a method for coding audio data comprising a sequence of digital audio samples, including the steps of:

i) multiplying the input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients.

The present invention also provides a method for coding audio data, including the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

determining a Fourier transform coefficient sequence as defined above;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients as defined above, so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

The present invention also provides a method for coding audio data including the steps of:

obtaining at least one input sequence of digital audio samples;

pre-processing the input sequence samples including applying a pre-multiplication factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence utilising a fast Fourier transform; and

post-processing the sequence of transform coefficients including applying first post-

multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

The present invention also provides a method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre-processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence utilising a fast Fourier transform; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

The present invention further provides A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples $x[n]$, $y[n]$ corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence $z[n]$, where $z[n] = x[n] + jy[n]$;

pre-processing the complex input sequence samples including applying a pre-multiplication factor $cos(\pi n/N) + jsin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,\ldots,(N-1)$;

transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k utilising a fast Fourier transform, wherein $k = 0,\ldots,(N/2-1)$; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0..N/2-1$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0..N/2-1$$

$$\begin{aligned} X_k = \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N) \\ - \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N) \end{aligned}$$

$$\begin{aligned} Y_k = \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N) \\ - \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N) \end{aligned}$$

where G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k) = \pi(2k+1)/4$.

The modified discrete cosine transform equation can be expressed as

$$X_k = \sum_{n=0}^{n=N-1} x[n] * \cos(2\pi*(2n+1)*(2k+1)/4N + \pi*(2k+1)/4) \qquad k=0...(N/2-1)$$

where *x[n]* is the input sequence for a channel and N is the transform length.

Instead of evaluating X_k in the form given above it could be computed as

$$\begin{aligned} X_k &= \cos\gamma*(g_{k,r}\cos(\pi(k+1/2)/N)-g_{k,i}\sin(\pi(k+1/2)/N)) \\ &\quad -\sin\gamma*(g_{k,r}\sin(\pi(k+1/2)/N)+g_{k,i}\cos(\pi(k+1/2)/N)) \\ &\quad g_{k,r},g_{k,i} \in \Re(\textit{set of real numbers}) \end{aligned}$$

where $G_k = g_{k,r} + jg_{k,i} = \sum_{n=0}^{n=N-1} (x[n]e^{j\pi n/N})*e^{j2\pi nk/N}$. The symbol j represents the imaginary number $\sqrt{-1}$. The expression $\sum_{n=0}^{n=N-1} (x[n]e^{j\pi n/N})*e^{j2\pi nk/N}$ is obtained from the well known FFT method, by first using transformation $x'[n]=x[n] * e^{j\pi n/N}$ and then computing the FFT $G_k = \sum_{n=0}^{n=N-1} x'[n]*e^{j2\pi nk/N}$.

For a two channel approach, a complex variable $z[n] = x[n]*e^{j\pi n/N} + jy[n]*e^{j\pi n/N}$ is defined, where $x[n]$ and $y[n]$ are sample sequence for the two channels and $e^{j\pi n/N}$ represents the pre-multiplication factor. Using FFT approach, the frequency coefficient Z_k for the variable $z[n]$ is computed. From Z_k the value $G_k = (Z_k + Z^*_{N-k-1})/2$ and $G'_k = (Z_k - Z^*_{N-k-1})/2j$, required to compute the final MDCT for each channel, respectively, is calculated.

If either or both the channels require short length transformers, two short transforms are taken using the above approach. If neither need short transform, a single long transform is used. As an additional step in reducing computation, the windowing function can be combined with the pre-processing stage.

Brief Description of the Drawings

The invention is described in detail hereinafter, by way of example only, with reference to preferred embodiments thereof and with aid of the accompanying drawings, wherein:

Figure 1 is a diagrammatic representation of a stream of audio data and the substructure arrangement thereof;

Figure 2 is a functional block diagram of a digital audio encoder;

Figure 3 is a functional block diagram of a system for encoding a single audio channel; and

Figure 4 is a functional block diagram of a system for encoding a pair of audio channels.

Detailed Description of the Preferred Embodiments

The above mentioned Advanced Television Systems Committee (ATSC) Standard document entitled "Digital Audio Compression (AC-3) Standard" (Document A/52, 20 December, 1995) describes methods for encoding and decoding audio signals, and is hereby expressly incorporated herein by reference.

In general, the input to an audio coder comprises a stream of digitised samples of the time domain analog signal. For a multi-channel encoder the stream consists of interleaved samples for each channel. The input stream is sectioned into blocks, each block containing N consecutive samples of each channel (see Fig. 1). Thus within a block the N samples of a channel form a sequence $\{x[0], x[1], x[2], \ldots, x[N\text{-}1]\}$.

The time domain samples are next converted to the frequency domain using an analysis filter bank (see Fig. 2). The frequency domain coefficients, thus generated, form a coefficient set which can be identified as $(X_0, X_1, X_2, \ldots, X_{N/2\text{-}1})$. Since the signal is real only the first $N/2$ frequency components are considered. Here X_0 is the lowest frequency (DC) component while $X_{N/2\text{-}1}$ is the highest frequency component of the signal.

Audio compression essentially entails finding how much of the information in the set (X_0, X_1, X_2, ..., $X_{N/2-1}$) is necessary to reproduce the original analog signal at the decoder with minimal audible distortion.

The coefficient set is normally converted into floating point format, where each coefficient is represented by an exponent and mantissa. The exponent set is usually transmitted in its original form. However, the mantissa is truncated to a fixed or variable number of decimal places. The value of number of bits for coding a mantissa is usually obtained from a bit allocation algorithm which for advanced psychoacoustic coders may be based on the masking property of the human auditory system. A low number of bits results in high compression ratio because less space is required to transmit the coefficients. However this causes very high quantization error leading to audible distortion. A good distribution of available bits to each mantissa forms the core of the most advanced encoders.

In some encoders such as the AC-3, the frequency domain transformation of signals is performed by the (MDCT) modified discrete cosine transform (Eq. 1).

$$X_k = \sum_{n=0}^{n=N-1} x[n] * \cos(2\pi*(2n+1)*(2k+1)/4N + \pi*(2k+1)/4) \quad k=0...(N/2-1) \quad \text{Eq. 1}$$

If directly implemented in the form given above, the MDCT requires $O(N^2)$ additions and multiplications.

Single Channel FFT

It is possible to reduce the number of required operations significantly if one is able to evaluate Eq. 1 using the well known Fast Fourier Transform method of J.W. Cooley and J.W. Tukey (1960). The general Discrete Fourier Transform (DFT) is given below (Eq. 2). It requires $O(N^2)$ complex additions and multiplications. By using the Fast Fourier Transform method the DFT in Eq. 2 can be computed with $O(N\log 2N)$ operations only.

$$F_k = \sum_{n=0}^{n=N-1} (x[n] * e^{2\pi jnk/N}) \qquad k=0..N-1 \qquad \text{Eq. 2}$$

Here j is the symbol for imaginary number, i.e. $j = \sqrt{-1}$.

Although it may not be immediately apparent how Eq. 1 can be transformed to Eq. 2, a careful analysis shows that this is indeed possible. To simplify Eq. 1, two functions can be defined

$$\alpha(n,k) = 2\pi(2n+1)(2k+1)/4N \qquad \text{Eq. 3}$$

$$\gamma(k) = \pi(2k+1)/4 \qquad \text{Eq. 4}$$

Then, using these functions, Eq. 1 can be rewritten as

$$X_k = \sum_{n=0}^{n=N-1} x[n] * \cos(\alpha(n,k) + \gamma(k)) \qquad \text{Eq. 5}$$

$$= \sum_{n=0}^{n=N-1} x[n] * (\cos\alpha(n,k)\cos\gamma(k) - \sin\alpha(n,k)\sin\gamma(k)) \qquad \text{Eq. 6}$$

In Eq. 6 the trigonometric equality, $\cos(a+b) = \cos a \cos b - \sin a \sin b$ is used for simplification. Furthermore, since the function $\gamma(k)$ is not dependant on variable n, it can be brought outside the summation expression to give

$$X_k = \cos\gamma(k) \sum_{n=0}^{n=N-1} x[n] * \cos\alpha(n,k) - \sin\gamma(k) \sum_{n=0}^{n=N-1} x[n] * \sin\alpha(n,k)$$

$$= T_1\cos\gamma(k) - T_2\sin\gamma(k) \qquad \text{Eq. 7}$$

where $T_1 = \sum_{n=0}^{n=N-1} x[n] * \cos\alpha(n,k)$ and $T_2 = \sum_{n=0}^{n=N-1} x[n] * \sin\alpha(n,k)$

The two terms, T_1 and T_2, can now be evaluated separately. Using Euler's identity $e^{i\theta} = \cos\theta + j\sin\theta$, we can express:

$\cos\alpha(n,k) = (e^{j\alpha(n,k)} + e^{-j\alpha(n,k)})/2$

and $\sin\alpha(n,k) = (e^{j\alpha(n,k)} - e^{-j\alpha(n,k)})/2j$.

Therefore we can rewrite the term T_1 as

$$T_1 = \sum_{n=0}^{n=N-1} x[n]*(e^{j\alpha}+e^{-j\alpha})/2 = 1/2\left(\sum_{n=0}^{n=N-1} x[n]*e^{j\alpha} + \sum_{n=0}^{n=N-1} x[n]*e^{-j\alpha}\right) \qquad \text{Eq. 8}$$
$$= 1/2\ (A_1 + A_2)$$

where $A_1 = \sum_{n=0}^{n=N-1} x[n]*e^{j\alpha}$ and $A_2 = \sum_{n=0}^{n=N-1} x[n]*e^{-j\alpha}$

Similarly

$$T_2 = \sum_{n=0}^{n=N-1} x[n]*(e^{j\alpha}-e^{-j\alpha})/2 = 1/2j\left(\sum_{n=0}^{n=N-1} x[n]*e^{j\alpha} - \sum_{n=0}^{n=N-1} x[n]*e^{-j\alpha}\right) \qquad \text{Eq. 9}$$
$$= 1/2j(A_1 - A_2)$$

The term A_1 can thus be evaluated from Eq. 8 and Eq. 9

$$\begin{aligned} A_1 &= \sum_{n=0}^{n=N-1} x[n]*e^{j\alpha} \\ &= \sum_{n=0}^{n=N-1} x[n]*e^{j(2\pi(2n+1)(2k+1)/4N)} \\ &= e^{j\pi(k+1/2)/N} * \sum_{n=0}^{n=N-1} (x[n]*e^{j\pi n/N}) * e^{j2\pi nk/N} \end{aligned} \qquad \text{Eq. 10}$$

If a complex variable is defined as:

$$x'[n]=x[n]*e^{j\pi n/N} \qquad \text{Eq. 11}$$

then Eq. 10 is simply:

$$\begin{aligned} A_1 &= e^{j\pi(k+1/2)/N} * \sum_{n=0}^{n=N-1} x'[n]*e^{j2\pi nk/N} \\ &= e^{j\pi(k+1/2)/N} * G_k \end{aligned} \qquad \text{Eq. 12}$$

where $G_k = \sum_{n=0}^{n=N-1} x'[n]*e^{j2\pi nk/N}$

The complex term $G_k = g_{k,r}+g_{k,i}$, where $g_{k,r}$ and $g_{k,i} \in \Re$ (set of real numbers) in Eq. 12 is essentially the same as F_k in Eq. 2. Therefore the FFT approach can be used to evaluate G_k. This brings down computation from $O(N^2)$ to $O(N\log N)$. Similarly, the second term A_2 in Eq. 8 and Eq. 9 can be evaluated

$$A_2 = \sum_{n=0}^{n=N-1} x[n]*e^{-j\alpha(n,k)} = e^{-j\pi(2k+1/2)/N} * \sum_{n=0}^{n=N-1} (x[n]*e^{-j\pi n/N})*e^{-j2\pi nk/N}$$
$$= e^{-j\pi(2k+1/2)/N} * G_k^* \qquad \text{Eq. 13}$$

where $G_k^* = \sum_{n=0}^{n=N-1} (x[n]*e^{-j\pi n/N})*e^{-j2\pi nk/N}$

Note that G_k^* is actually the complex conjugate of G_k which was obtained by Eq. 12. That is, if $G_k = g_{k,r}+g_{k,i}$, where $g_{k,r}$ and $g_{k,i} \in \Re$ as defined earlier, then $G_k^* = g_{k,r} - jg_{k,i}$. Therefore G_k^* in Eq. 13 does not need to be computed again, and the result from Eq. 12 can be re-used. That is, only one FFT needs to be computed for the evaluation of T_1. The result of Eq. 8 to Eq. 13 is thus

$$T_1 = 1/2(e^{j\pi(k+1/2)/N}\, G_k + e^{-j\pi(k+1/2)/N}\, G_k^*) \qquad \text{Eq. 14}$$

Next, the term T_2 can be analysed

$$T_2 = \sum_{n=0}^{n=N-1} x[n]*(e^{j\alpha}-e^{-j\alpha})/2j$$
$$= 1/2j(A_1-A_2)$$
$$= 1/2j(e^{j\pi(k+1/2)/N}\, G_k - e^{-j\pi(k+1/2)/N}\, G_k^*) \qquad \text{Eq. 15}$$

Finally, after simplifications of Eq. 7, 14 and 15

$$\begin{aligned} X_k &= \cos\gamma(k)\ 1/2(e^{j\pi(k+1/2)/N}\,G_k + e^{-j\pi(k+1/2)/N}\,G_k^*) \\ &\quad - \sin\gamma(k)\ 1/2j(e^{j\pi(k+1/2)/N}\,G_k - e^{-j\pi(k+1/2)/N}\,G_k^*) \\ &= \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N) \\ &\quad - \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N) \\ &= \cos\gamma * T_1 - \sin\gamma * T_2 \end{aligned} \qquad \text{Eq. 16}$$

The term $G_k = g_{k,r} + jg_{k,I}$ is computed in $O(N\log N)$ operation by use of FFT algorithms. The additional operation outlined in Eq. 16 to extract the final X_k is only of order $O(N)$. Therefore the MDCT can now be computed in $O(N\log_2 N)$ time. The operations required to obtain the MDCT are illustrated in Fig. 3.

Combining Two Channels into Single FFT

Suppose the multi-channel encoder is required to process m audio channels. Instead of computing an FFT for each channel as described in the previous section, it is possible to further reduce the computational requirement of the coder by combining two channels and using a single FFT only. In effect, instead of m FFTs only $m/2$ FFTS need to be computed.

If the input sequence are real numbers then it is known that DFT for any two channels can be computed with only one FFT block by considering the input as a complex number. The real part is formed from the sequence for any one channel and the imaginary part is from data of another channel. After the Fourier Transform is computed for the resulting complex variable, the resulting transform for each channel can be easily retrieved.

However, in the present case the input data to the FFT block is actually a complex number (formed by multiplying the real data by complex variable $e^{j\pi n/N}$). In this case, there is no straightforward way of retrieving the frequency transform after having combined two channels. However, using some processing after the FFT one can still compute the DFT of two channel using a single FFT block.

Let $\{x[0],x[1],x[2],\ldots,x[N-1]\}$ be N input samples of the first channel and $\{y[0],y[1],y[2],\ldots y[N-1]\}$ be the samples for the second channel. As described above, the frequency coefficients $G_k = \sum_{n=0}^{n=N-1} x[n]e^{j\pi n/N} * e^{j2\pi nk/N}$ (Eq. 12 and 13) must be obtained for the first channel; and similarly, for the second channel

$$G'_k = \sum_{n=0}^{n=N-1} y[n]e^{j\pi n/N} * e^{j2\pi nk/N}$$

Defining complex variable $z[n] = x[n]*e^{j\pi n/N} + jy[n]*e^{j\pi n/N}$ Eq. 17

and computing its DFT using the FFT method, yields

$$\begin{aligned} Z_k &= \sum_{n=0}^{n=N-1} z[n]*e^{j2\pi nk/N} \qquad k=0..N-1 \\ &= \sum_{n=0}^{n=N-1} (x[n] + jy[n])e^{j\pi n/N}*e^{j2\pi nk/N} \\ &= \sum_{n=0}^{n=N-1} (x[n] + jy[n])*e^{j2\pi n(k+1/2)/N} \end{aligned}$$ Eq. 18

Now substituting N-k for k in the above expression,

$$\begin{aligned} Z_{N-k} &= \sum_{n=0}^{n=N-1} (x[n] + jy[n])*e^{j2\pi n(N-k+1/2)/N} \qquad k=0..N-1 \\ &= \sum_{n=0}^{n=N-1} (x[n] + jy[n])*e^{j2\pi n(-k+1/2)/N}*e^{-j2\pi n} \\ &= \sum_{n=0}^{n=N-1} (x[n] + jy[n])*e^{j2\pi n(-k+1/2)/N} \end{aligned}$$ Eq. 19

Since $e^{j2\pi n} = 1$, $n \in I$ (the set of integers), the term $e^{j2\pi n}$ vanishes in the above expression. Taking the complex conjugate of Z_{N-k}:

$$\begin{aligned} Z^*_{N-k} &= \sum_{n=0}^{n=N-1} (x[n]-jy[n])*e^{-j2\pi n(-k+1/2)/N} \\ &= \sum_{n=0}^{n=N-1} (x[n]-jy[n])*e^{j2\pi n(k-1/2)/N} \end{aligned}$$ Eq. 20

Using Eq. 18 and 20, separate expressions for G_k and G'_k are required. In a simple case the conjugates in Eq. 18 and 20 should add and subtract to give the required expressions. However in this instance that is not the case. But, substituting *N-k* by *N-k-1* in Eq. 18, the following is obtained

$$Z^*_{N-k-1} = \sum_{n=0}^{n=N-1} (x[n] - jy[n]) * e^{j2\pi n(k+1/2)/N} \qquad \text{Eq. 21}$$

Now the term $e^{j2\pi n(k+1/2)/N}$ is common in both Eq. 17 and 19, and it is possible to isolate.

$$\begin{aligned} Z_k + Z^*_{N-k-1} &= \sum_{n=0}^{n=N-1} x[n] * e^{j2\pi n(k+1/2)/N} + j \sum_{n=0}^{n=N-1} y[n] * E^{j2\pi n(k+1/2)/N} \\ &+ \left(\sum_{n=0}^{n=N-1} x[n] * e^{j2\pi n(k+1/2)/N} - j \sum_{n=0}^{n=N-1} y[n] * E^{j2\pi n(k+1/2)/N} \right) \\ &= 2 \sum_{n=0}^{n=N-1} (x[n] e^{j\pi n/N}) * e^{j2\pi nk/N} \\ &= 2G_k \end{aligned}$$

Similarly,

$$\begin{aligned} Z_k - Z^*_{N-k-1} &= \sum_{n=0}^{n=N-1} x[n] * e^{j2\pi n(k+1/2)/N} + j \sum_{n=0}^{n=N-1} y[n] * E^{j2\pi n(k+1/2)/N} \\ &- \left(\sum_{n=0}^{n=N-1} x[n] * e^{j2\pi n(k+1/2)/N} - j \sum_{n=0}^{n=N-1} y[n] * E^{j2\pi n(k+1/2)/N} \right) \\ &= 2j \sum_{n=0}^{n=N-1} (y[n] e^{j\pi n/N}) * e^{j2\pi nk/N} \\ &= 2jG'_k \end{aligned}$$

That is

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0..N/2-1 \qquad \text{Eq. 22}$$

and

$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0..N/2-1 \qquad \text{Eq. 23}$$

From the expression from Eq. 22 and 23 into Eq. 16, the MDCT for each channel is obtained. The overall process is illustrated in Fig. 4.

Transform Length Adjustment Technique

The frequency transform length N is decided by the encoder based on temporal and spectral resolution requirements. The input signal is usually analysed with a high frequency bandpass filter to detect the presence of transients. This information is used to adjust the block length, restricting quantization noise associated with the transient within a small temporal region about the transient, avoiding temporal masking. Thus, if transient is detected in a channel, two short transform of length $N/2$ each are taken. In the absence of transient, a single long transform of length N is used, thus providing higher spectral resolution.

From the method described in the previous section for computing MDCT for two channels using a single FFT block, it is evident that the transform length for the two paired channels must be the same. Therefore, pairing for the transformation phase much be such that channels with identical transform length are grouped together.

It is however possible that not all channels can be paired with such convenience. Assume that the total number of channels are an even number (if not, take a single FFT for one channel and the rest form an even group). Suppose out of the m channels, l need long transform and therefore $m-l$ require short transform.

If l is an even number, then since the total is even, it follows that $l-m$ is also even. In this case, from the l channels that need long transform, $l/2$ pairs are formed and for each of the $l/2$ pairs a single FFT is computed to estimate the MDCT for the original paired channels. Similarly, the $l-m$ channels are paired to form $(l-m)/2$ pairs and for the $(l-m)/2$ pairs two short FFTs are computed.

Now consider the case when 1 $l = 2r + 1$ is an odd number. Therefore $m - l = 2s + 1$ is

also an odd number. The $2r$ channels requiring long transform are paired together to form r pairs and then $2r$ transforms are computed using r FFTs only. Similarly, for the $2s$ channels s pairs are formed. What remains is one channel requiring long transform and another requiring two short transforms. Both of these channels are paired together and two short FFTs are computed to derive the MDCT.

The rationale for constraining the long transform to two short ones is as follows. A short transform is required for restricting quantization noise associated with the transient within a small temporal region about the transient, avoiding temporal masking. A long transform gives slight better frequency resolution but the error is not much compared to the case when in the presence of transient a long transform is utilised. Forcing a long transform onto a channel in the presence of transient leads to greater distortion in the final produced music. This conjecture was proven true by experimental studies on benchmark music streams.

Combining Windowing with pre-processing

Before the time domain signal $x[n]$ is transformed to the frequency domain, a windowing function is usually applied. Thus, if the sampled signal is $p[n]$ then the sequence that is applied to the frequency transformation block is $x[n] = p[n] * w[n]$, where $w[n]$ is the windowing function. From the previous sections we noted that before the FFT is computed for a block a pre-processing is performed as given in Eq. 11 (reproduced below for convenience). Thus

$$\begin{aligned} x'[n] &= x[n] * e^{j\pi n/N} \\ &= (p[n] * w[n]) * e^{j\pi n/N} \\ &= (p[n] * w[n]) * (\cos \pi n/N + j \sin \pi n/N) \\ &= p[n] * ((w[n] * \cos \pi n/N) + j(w[n] * \sin \pi n/N)) \qquad \text{Eq. 24} \end{aligned}$$

From Eq. 24 we note that the windowing function can be combined with the cosine and sine multiplication required in Eq. 11. This brings down the computation even further since the sine and cosine are usually implemented in a real time system as table-lookup. If

two tables are constructed as defined below

$$rcos[n] = w[n] * \cos(\pi n/N)$$
$$rsin[n] = w[n] * \sin(\pi n/N)$$

then Eq. 11 can be rewritten as

$$x'[n] = (p[n] * rcos[n]) + j(p[n] * r\sin[n]) \qquad \text{Eq. 25}$$

Although the invention has been described herein primarily in terms of its mathematical derivation and application, and the procedures required for implementation, it will be readily recognised by those skilled in the art that the procedures described can be implemented by means of any desired computational apparatus. For example, the invention may be embodied in computer software operating on general purpose computing equipment, or may be embodied in purpose built circuitry or contained in microcode or the like in an integrated circuit or set of integrated circuits.

The foregoing detailed description of embodiments of the invention has been presented by way of example only, and is not intended to be considered limiting to the invention as defined in the claims appended hereto.

Glossary of Equations:

MDCT

$$X_k = \sum_{n=0}^{n=N-1} x[n] * \cos(2\pi*(2n+1)*(2k+1)/4N + \pi*(2k+1)/4) \qquad k=0...(N/2-1)$$
$$= \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N)$$
$$- \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N)$$
$$= T_1\cos\gamma(k) - T_2\sin\gamma(k)$$

$$T_1 = \sum_{n=0}^{n=N-1} x[n]*\cos\alpha(n,k) \qquad T_2 = \sum_{n=0}^{n=N-1} x[n]*\sin\alpha(n,k)$$
$$= 1/2(A_1 + A_2) \qquad = 1/2j(A_1 - A_2)$$

$$A_1 = \sum_{n=0}^{n=N-1} x[n]*e^{j\alpha} \qquad A_2 = \sum_{n=0}^{n=N-1} x[n]*e^{-j\alpha}$$
$$= e^{j\pi(k+1/2)/N}*G_k \qquad = e^{-j\pi(2k+1/2)/N} * G_k^*$$

$$G_k = \sum_{n=0}^{n=N-1} (x[n]*e^{j\pi n/N})*e^{j2\pi nk/N} \qquad G_k^* = \sum_{n=0}^{n=N-1} (x[n]*e^{-j\pi n/N})*e^{-j2\pi nk/N}$$

$$T_1 = 1/2(e^{j\pi(k+1/2)/N}\, G_k + e^{-j\pi(k+1/2)/N}\, G_k^*)$$

$$T_2 = 1/2j(e^{j\pi(k+1/2)/N}\, G_k - e^{-j\pi(k+1/2)/N}\, G_k^*)$$

$$G_k = (Z_k + Z_{N-k-1}^*)/2 \qquad k=0..N/2-1$$

$$G'_k = (Z_k - Z_{N-k-1}^*)/2j \qquad k=0..N/2-1$$

$$\alpha(n,k) = 2\pi(2n+1)(2k+1)/4\mathrm{N}$$
$$\gamma(k) = \pi(2k+1)/4$$

Claims

1. A method for coding audio data comprising a sequence of digital audio samples, including the steps of:

i) multiplying the input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

iii) for each transform coefficient in the sequence, multiplying the real and imaginary components of the transform coefficient by respective second trigonometric function factors, adding the multiplied real and imaginary transform coefficient components to generate an addition stream coefficient, and subtracting the multiplied real and imaginary transform coefficient components to generate a subtraction stream coefficient;

iv) multiplying the addition and subtraction stream coefficients with respective third trigonometric function factors; and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients to generate audio coded frequency domain coefficients.

2. A method for coding audio data as claimed in claim 1, wherein the audio coded frequency domain coefficients comprise modified discrete cosine transform coefficients.

3. A method for coding audio data as claimed in claim 1 or 2, wherein the first trigonometric function factor for each audio sample is a function of the audio sample sequence position and the number of samples in the sequence.

4. A method for coding audio data as claimed in claim 3, wherein the respective second trigonometric function factors for each transform coefficient in the sequence are respective functions of the transform coefficient sequence position and the number of coefficients in the sequence.

5. A method for coding audio data as claimed in claim 4, wherein the respective third trigonometric function factors are respective functions of the transform coefficient sequence position.

6. A method for coding audio data as claimed in claim 5, wherein step i) comprises multiplying the input sequence samples $x[n]$ by the first trigonometric function factor $cos(\pi n/N)$ to generate the intermediate sample sequence, where:

$x[n]$ are the input sequence audio samples;

N is the number of input sequence audio samples; and

$n = 0,\ldots,N-1$.

7. A method for coding audio data as claimed in claim 6, wherein step ii) comprises computing the fast Fourier transform of the intermediate sample sequence so as to generate said transform coefficient sequence $G_k = g_{k,r} + jg_{k,i}$, where:

G_k is the transform coefficient sequence;

$g_{k,r}$ are the real transform coefficient components;

$g_{k,i}$ are the imaginary transform coefficient components; and

$k = 0,\ldots,(N/2-1)$.

8. A method for coding audio data as claimed in claim 7, wherein step iii) comprises determining the addition stream coefficients T_2 and subtraction stream coefficients T_1 according to:

$$T_1 = g_{k,r}\, cos(\pi(k+1/2)/N) - g_{k,i}\, sin(\pi(k+1/2)/N)$$

$$T_2 = g_{k,r}\, cos(\pi(k+1/2)/N) + g_{k,i}\, sin(\pi(k+1/2)/N)$$

where T_1 and T_2 are the subtraction stream and addition stream coefficients, respectively.

9. A method for coding audio data as claimed in claim 8, wherein steps iv) and v) comprise generating the audio coded frequency domain coefficients X_k according to:

$$X_k = T_1\, cos(\pi(2k+1)/4) - T_2\, sin(\pi(2k+1)/4)$$

where X_k are the audio coded frequency domain coefficients; and

$cos(\pi(2k+1)/4)$ and $sin(\pi(2k+1)/4)$ are the third trigonometric function factors.

10. A method for coding audio data, including the steps of:

combining first and second sequences of digital audio samples from first and second audio channels into a single complex sample sequence;

determining a Fourier transform coefficient sequence as defined in any preceding claim;

generating first and second transform coefficient sequences by combining and/or differencing first and second selected transform coefficients from said Fourier transform coefficient sequence; and

for each of the first and second transform coefficient sequences, generating audio coded frequency domain coefficients as defined in any preceding claim, so as to generate respective sequences of said audio coded frequency domain coefficients for the first and second audio channels.

11. A method for coding audio data as claimed in claim 10, wherein the step of generating first and second transform coefficient sequences comprises, for each corresponding coefficient in the first and second transform coefficient sequences, selecting first and second transform coefficients from said Fourier transform coefficient sequence, determining a complex conjugate of said second transform coefficient, combining said first transform coefficient and said complex conjugate for said first transform coefficient sequence and differencing said first transform coefficient and said complex conjugate for said second transform coefficient sequence.

12. A method for coding audio data as claimed in claim 10 or 11, wherein the multiplying step i) comprises multiplying the input sequence samples $z[n]$ by the first trigonometric function factor $cos(\pi n/N) + jsin(\pi n/N)$ to generate the intermediate sample sequence, where:

$z[n] = x[n] + jy[n]$ is the complex sample sequence;

$x[n]$ is the first sequence of digital audio samples;

$y[n]$ is the second sequence of digital audio samples;

N is the number of input sequence audio samples in each sequence;

$n = 0, \ldots, N-1$; and

j is the complex constant.

13. A method for coding audio data as claimed in claim 11 or 12, wherein said first and second transform coefficient sequences are generated according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j$$

where G_k is said first transform coefficient sequence;

G'_k is said second transform coefficient sequence;

N is the number of input sequence audio samples;

$k = 0,\ldots,(N/2-1)$;

Z_k is said first transform coefficient;

Z^*_{N-k-1} is the complex conjugate of said second transform coefficient; and

j is the complex constant.

14. A method for coding audio data as claimed in any one of claims 10 to 13, including examining said first and second sequences of digital audio samples to determine a short or long transform length, and coding the audio samples using a short or long transform length as determined.

15. A method for coding audio data comprising sequences of digital audio samples from a plurality of audio channels, comprising determining a transform length for each of the channels, pairing the channels according to their determined transform length, and coding the audio samples of first and second channels in each pair, as defined in any one of claims 10 to 13, according to the determined transform length.

16. A method for coding audio data as claimed in any preceding claim, including applying a windowing function in combination with said multiplying step i).

17. A method for coding audio data including the steps of:

obtaining at least one input sequence of digital audio samples;

pre-processing the input sequence samples including applying a pre-multiplication

factor to obtain modified input sequence samples;

transforming the modified input sequence samples into a transform coefficient sequence utilising a fast Fourier transform; and

post-processing the sequence of transform coefficients including applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

18. A method as claimed in claim 17, wherein the pre-multiplication factor, and first and second post-multiplication factors are trigonometric function factors.

19. A method as claimed in claim 18, wherein the pre-multiplication factor applied to each digital audio sample in the input sequence is a trigonometric function of the audio sample sequence position and the number of samples in the sequence.

20. A method as claimed in claim 18, wherein the first post-multiplication factors for each transform coefficient in the sequence are trigonometric functions of the transform coefficient sequence position and the number of coefficients in the sequence.

21. A method as claimed in claim 18, wherein the second post-multiplication factor for each difference or combination result is trigonometric functions of the transform coefficient sequence position of the coefficients used in the difference or combination.

22. A method as claimed in any one of claims 17 to 21, wherein the pre-processing operations are performed on each sample in the input sequence individually.

23. A method as claimed in any one of claims 17 to 22, wherein the post-processing operations are performed on each transform coefficient in the sequence individually.

24. A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence;

pre-processing the complex input sequence samples including applying a pre-multiplication factor to obtain modified complex input sequence samples;

transforming the modified complex input sequence samples into a complex transform coefficient sequence utilising a fast Fourier transform; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels including, for each corresponding frequency domain coefficient in the first and second sequences, selecting first and second complex transform coefficients from said sequence of complex transform coefficients, combining the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said first channel and differencing the first complex transform coefficient and the complex conjugate of the second complex transform coefficient for said second channel, and applying respective post-multiplication factors to the combination and difference to obtain said audio coded frequency domain coefficients corresponding to the first and second audio channels.

25. A method as claimed in claim 24, wherein the pre-multiplication factor for each sample in the complex input sample sequence comprises a complex trigonometric function of the complex input sample sequence position and the number of samples in the sequence.

26. A method as claimed in claim 24 or 25, wherein the post-processing for each of the first and second channels includes applying first post-multiplication factors to the real and imaginary coefficient components, differencing and combining the post-multiplied real and imaginary components, applying second post-multiplication factors to the difference and combination results, and differencing to obtain a sequence of modified discrete cosine transform coefficients representing said input sequence of digital audio samples.

27. A method for coding audio data including the steps of:

obtaining first and second input sequences of digital audio samples $x[n]$, $y[n]$ corresponding to respective first and second audio channels;

combining the first and second input sequences of digital audio samples into a single complex input sample sequence $z[n]$, where $z[n] = x[n] + jy[n]$;

pre-processing the complex input sequence samples including applying a pre-multiplication factor $cos(\pi n/N) + jsin(\pi n/N)$ to obtain modified complex input sequence samples, where N is the number of audio samples in each of the first and second input sequences and $n = 0,\ldots,(N\text{-}1)$;

transforming the modified complex input sequence samples into a complex transform coefficient sequence Z_k utilising a fast Fourier transform, wherein $k = 0,\ldots,(N/2\text{-}1)$; and

post-processing the sequence of complex transform coefficients to obtain first and second sequences of audio coded frequency domain coefficients corresponding to the first and second audio channels X_k, Y_k according to:

$$G_k = (Z_k + Z^*_{N-k-1})/2 \qquad k=0..N/2-1$$

$$G'_k = (Z_k - Z^*_{N-k-1})/2j \qquad k=0..N/2-1$$

$$\begin{aligned} X_k = {} & \cos\gamma * (g_{k,r}\cos(\pi(k+1/2)/N) - g_{k,i}\sin(\pi(k+1/2)/N) \\ & - \sin\gamma * (g_{k,r}\sin(\pi(k+1/2)/N) + g_{k,i}\cos(\pi(k+1/2)/N) \end{aligned}$$

$$\begin{aligned} Y_k = {} & \cos\gamma * (g'_{k,r}\cos(\pi(k+1/2)/N) - g'_{k,i}\sin(\pi(k+1/2)/N) \\ & - \sin\gamma * (g'_{k,r}\sin(\pi(k+1/2)/N) + g'_{k,i}\cos(\pi(k+1/2)/N) \end{aligned}$$

where G_k is a transform coefficient sequence for the first channel;

G'_k is a transform coefficient sequence for the second channel;

$g_{k,r}$ and $g_{k,i}$ are the real and imaginary transform coefficient components of G_k;

$g'_{k,r}$ and $g'_{k,i}$ are the real and imaginary transform coefficient components of G'_k;

Z^*_{N-k-1} is the complex conjugate of Z_{N-k-1}; and

$\gamma(k) = \pi(2k+1)/4$.

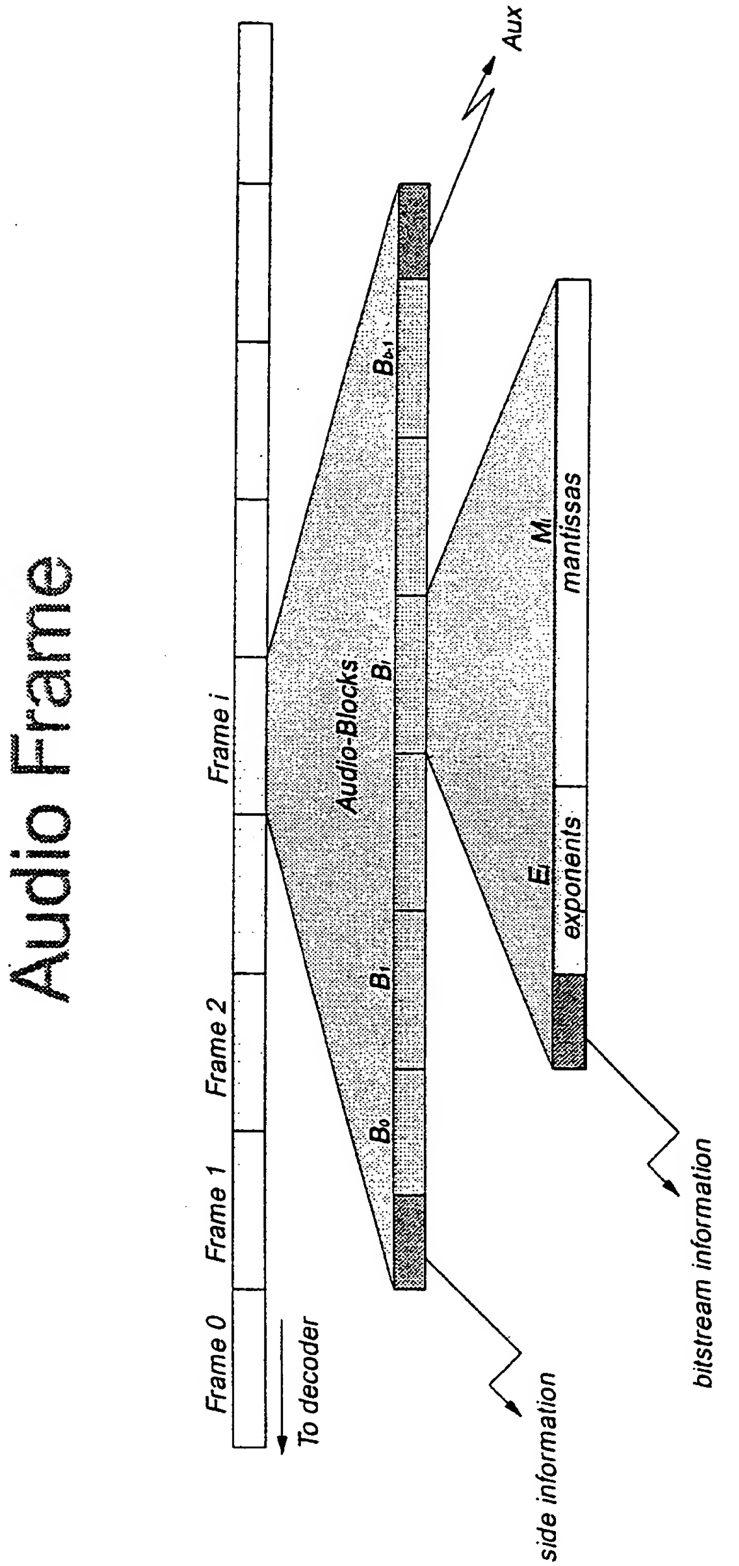
Audio Frame
Frame 0
Frame 1
Frame 2
Frame i
To decoder
Audio-Blocks
B_0
B_1
B_i
B_{b-1}
Aux
side information
E_i
exponents
M_i
mantissas
bitstream information

Fig. 1

2/4

AUDIO ENCODER

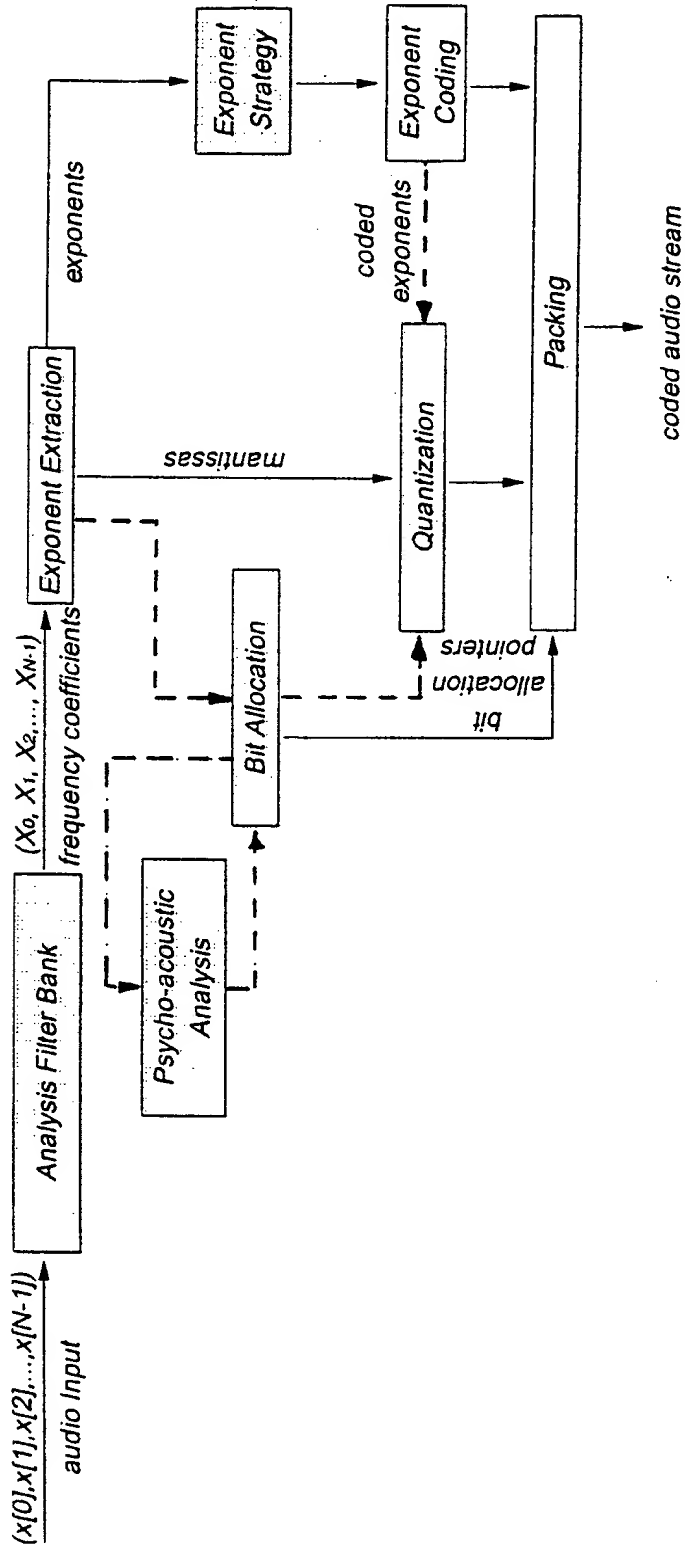

Fig. 2

Fast Modified Discrete Cosine Transform (single channel)

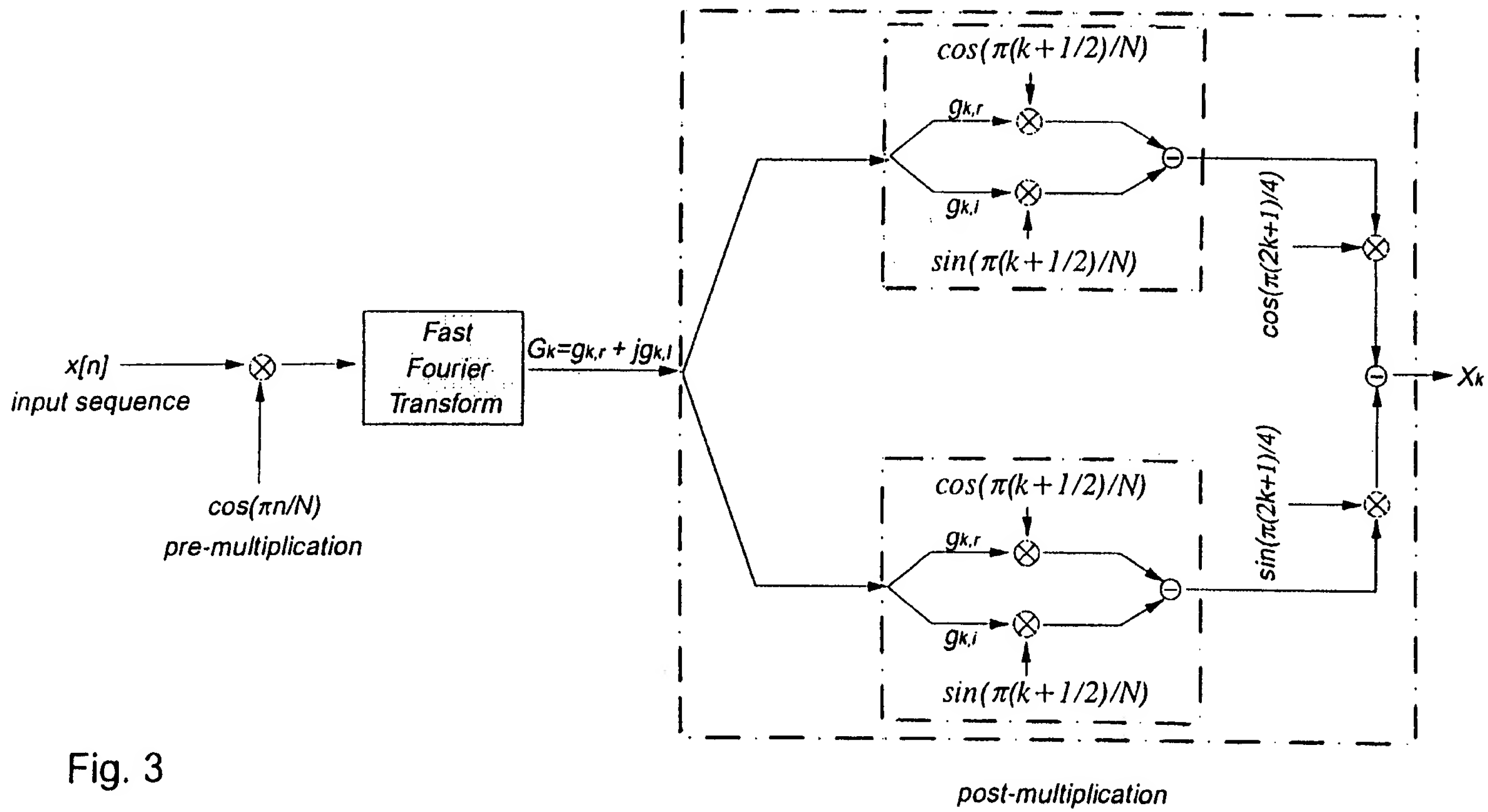

Fig. 3

Combined Fast Modified
Discrete Cosine Transform
(two channels)

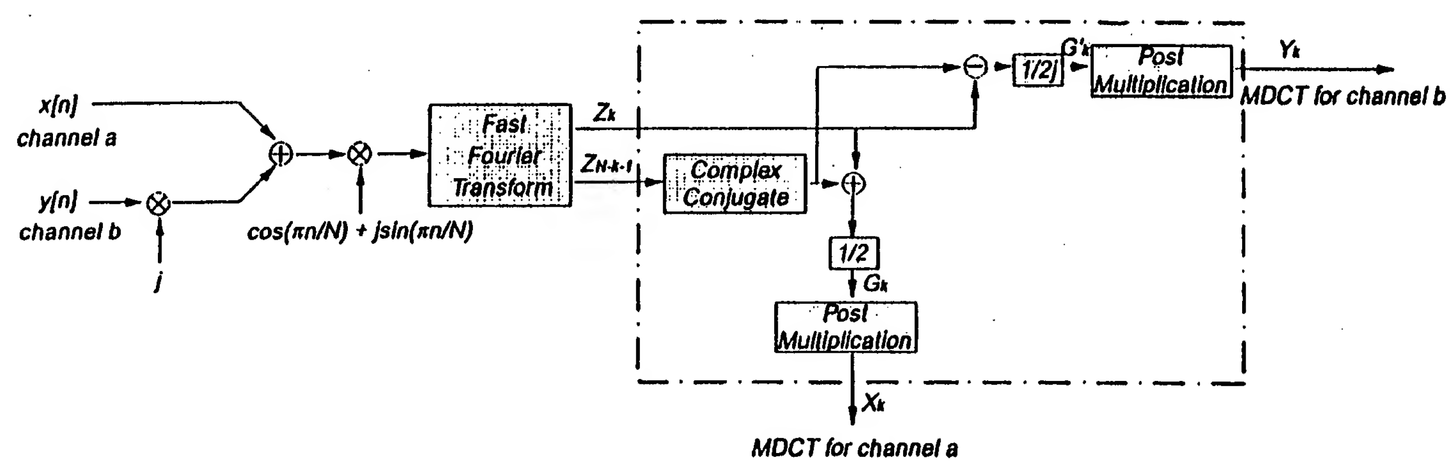

Fig. 4

INTERNATIONAL SEARCH REPORT

International Application No
PCT/SG 98/00014

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04H1/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04H

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP 0 506 111 A (MITSUBISHI ELECTRIC CORP) 30 September 1992 see page 2, line 1 - page 5, line 16; claim 1; figure 1	1,10,17,24,27
A	EP 0 590 790 A (SONY CORP) 6 April 1994 see page 2, line 1 - page 6, line 11; claims 1,8; figure 1	1,10,17,24,27
A	US 5 181 183 A (MIYAZAKI TAKASHI) 19 January 1993 see column 1, line 1 - column 2, line 27; claim 1; figure 1	1,10,17,24,27

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"A" document defining the general state of the art which is not considered to be of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
13 November 1998	23/11/1998

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	De Haan, A.J.

1

INTERNATIONAL SEARCH REPORT

International Application No
PCT/SG 98/00014

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	EP 0 564 089 A (AMERICAN TELEPHONE & TELEGRAPH) 6 October 1993 see page 2, line 1 - page 3, line 57; claim 1; figure 1 ---	1,10,17, 24,27
A	US 5 592 584 A (FERREIRA ANIBAL J ET AL) 7 January 1997 see column 1, line 1 - column 3, line 67; claim 1; figures 1,2 ---	1,10,17, 24,27
A	EP 0 718 746 A (PHILIPS ELECTRONIQUE LAB ;PHILIPS ELECTRONICS NV (NL)) 26 June 1996 see page 2, line 1 - page 3, line 3; claim 1; figure 1 -----	1,10,17, 24,27

1

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/SG 98/00014

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0506111 A	30-09-1992	JP 4313157 A	05-11-1992
		US 5249146 A	28-09-1993
EP 0590790 A	06-04-1994	JP 6112909 A	22-04-1994
		US 5646960 A	08-07-1997
		US 5640421 A	17-06-1997
US 5181183 A	19-01-1993	JP 2646778 B	27-08-1997
		JP 3211604 A	17-09-1991
EP 0564089 A	06-10-1993	CA 2090052 A	03-09-1993
		JP 6029859 A	04-02-1994
		US 5592584 A	07-01-1997
US 5592584 A	07-01-1997	CA 2090052 A	03-09-1993
		EP 0564089 A	06-10-1993
		JP 6029859 A	04-02-1994
EP 0718746 A	26-06-1996	JP 8241187 A	17-09-1996
		US 5684730 A	04-11-1997

APPENDIX E

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

WITH ARTICLE 34 AMENDMENTS

(COPY FILED WITH U.S. NATIONAL PHASE APPLICATION ON AUGUST 18, 2000)

PATENT COOPERATION TREATY

PCT

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference SGS/50566	FOR FURTHER ACTION See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)	
International application No. PCT/SG98/00014	International filing date *(day/month/year)* 21/02/1998	Priority date *(day/month/year)* 21/02/1998
International Patent Classification (IPC) or national classification and IPC H04H1/00		
Applicant SGS-THOMSON MICROELECTRONICS ASIA PACIFIC et al.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 6 sheets, including this cover sheet.

   ☒ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of 2 sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations suporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☒ Certain observations on the international application

Date of submission of the demand	Date of completion of this report
19/08/1999	26.05.00
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer Snell, T Telephone No. +49 89 2399 8802

Form PCT/IPEA/409 (cover sheet) (January 1994)

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

1,3-18	as originally filed			
2	as received on	04/01/2000	with letter of	21/12/1999

Claims, No.:

1-27	as originally filed

Drawings, sheets:

1/4,2/4,4/4	as originally filed			
3/4	as received on	04/01/2000	with letter of	21/12/1999

2. The amendments have resulted in the cancellation of:

   ☐ the description, pages:
   ☐ the claims, Nos.:
   ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes:	Claims	1-27
	No:	Claims	
Inventive step (IS)	Yes:	Claims	1-27
	No:	Claims	
Industrial applicability (IA)	Yes:	Claims	1-27
	No:	Claims	

2. Citations and explanations

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

VIII. Certain observations on the international application

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

see separate sheet

Cited documents

D1: EP-A-0564089

Re Item V

Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. The invention relates to a method for coding audio signals. According to the closest prior art D1 it is known to code an audio signal by performing a modified discrete cosine transform (MDCT) by utilising a fast Fourier transform (FFT).

2. The present invention, as defined in claim 1, also utilises the principle of utilising a FFT to perform a MDCT. In contrast to D1, however, which only processes the real component of the Fourier transform samples, in the present invention real and imaginary components are processed followed by subtraction and addition functions to derive the output frequency domain coefficients. No other available document gives any hint to modify D1 to produce the present invention as defined in claim 1, so that an inventive step is acknowledged. The same applies to claim 17, a further independent claim directed at the same embodiment.

3. Independent claims 10 and 24 concern a further embodiment based on the same inventive principle, but modified to combine two channels in a single FFT. Claim 27 concerns the same embodiment as claim 24 but expressed in full mathematical detail.

   Claims 1, 10, 17, 24 and 27 therefore meet the requirements for novelty and inventive step (Articles 33(1)-(3) PCT).

4. Claims 2-9, 11-16, 18-23, 25 and 26 are dependent on either claim 1 or 24 and therefore also meet the requirements for novelty and inventive step (Articles 33(1)-(3) PCT).

Re Item VII

Certain defects in the international application

1. The phrase "hereby expressly incorporated herein by reference" on page 10 should have been deleted as the application should be self-contained; such referenced documents are not regarded as part of the disclosure unless they contain matter not included in the application which is essential to the invention, in which case the subject-matter in question would have had to be incorporated into the description. This however is not the case here (see PCT Guidelines II-4.17).

Re Item VIII

Certain observations on the international application

1. Although claims 1 and 17 have been drafted as separate independent claims, they appear to relate effectively to the same subject-matter (ie the embodiment of figure 3) and to differ from each other only with regard to the definition of the subject-matter for which protection is sought and/or in respect of the terminology used for the features of that subject-matter. The aforementioned claims therefore lack conciseness. Moreover, lack of clarity of the claims as a whole arises, since the plurality of independent claims makes it difficult, if not impossible, to determine the matter for which protection is sought, and places an undue burden on others seeking to establish the extent of the protection.

   Hence, claims 1 and 17 do not meet the requirements of Article 6 PCT.

2. The same objection applies to claims 10 and 24, which relate to the embodiment of figure 4 (Article 6 PCT).

3. Moreover, the formulation of claim 10 lacks clarity as it includes references to individual features of previous claims rather than to the whole of said claims, leading to obscurity in construing the exact scope of protection (Article 6 PCT). However, if claim 10 had been clarified to explicitly define all steps rather than rely on such references, it would be the same as claim 24. Claim 10 is therefore superfluous.

4. In view of the above, only independent claims 1, 24 and 27 should have been retained, with dependent claims as appropriate.

The exponents are usually transmitted in their original form. However, each mantissa must be truncated to a fixed or variable number of decimal places. The number of bits to be used for coding each mantissa is obtained from a bit allocation algorithm which may be based on the masking property of the human auditory system. Lower numbers of bits result in higher compression ratios because less space is required to transmit the coefficients. However, this may cause high quantization errors, leading to audible distortion. A good distribution of available bits to each mantissa forms the core of the advanced audio coders.

The frequency transformation phase has one of the greatest computation requirements in a transform coder. Therefore, an efficient implementation of this phase can decrease the computation requirement of the system significantly and make real time operation of the encoder more easily attainable.

In some encoders such as those specified in the AC-3 standard, the frequency domain transformation of signals is performed by the modified discrete cosine transform (MDCT). If directly implemented, the MDCT requires $O(N^2)$ additions and multiplications. However it has been found possible to reduce the number of required operations significantly if the MDCT equation is able to be computed in a from that is amenable to the use of the well known Fast Fourier Transform (FFT) method of J.W. Cooley and J.W. Tukey (1960). A known application of an FFT to an MDCT is disclosed in, for example, EP-A-0564089.

The present invention seeks to provide an alternative computation method using a Fast Fourier Transform. Moreover, the invention seeks to use a single FFT for two channels to achieve greater reduction in computational requirements of the system.

Summary of the Invention

In accordance with the present invention there is provided a method for coding audio data comprising a sequence of digital audio samples, including the steps of:

i) multiplying the input samples with a first trigonometric function factor to generate an intermediate sample sequence;

ii) computing a fast Fourier transform of the intermediate sample sequence to generate a Fourier transform coefficient sequence;

AMENDED SHEET

Fast Modified Discrete Cosine Transform (single channel)

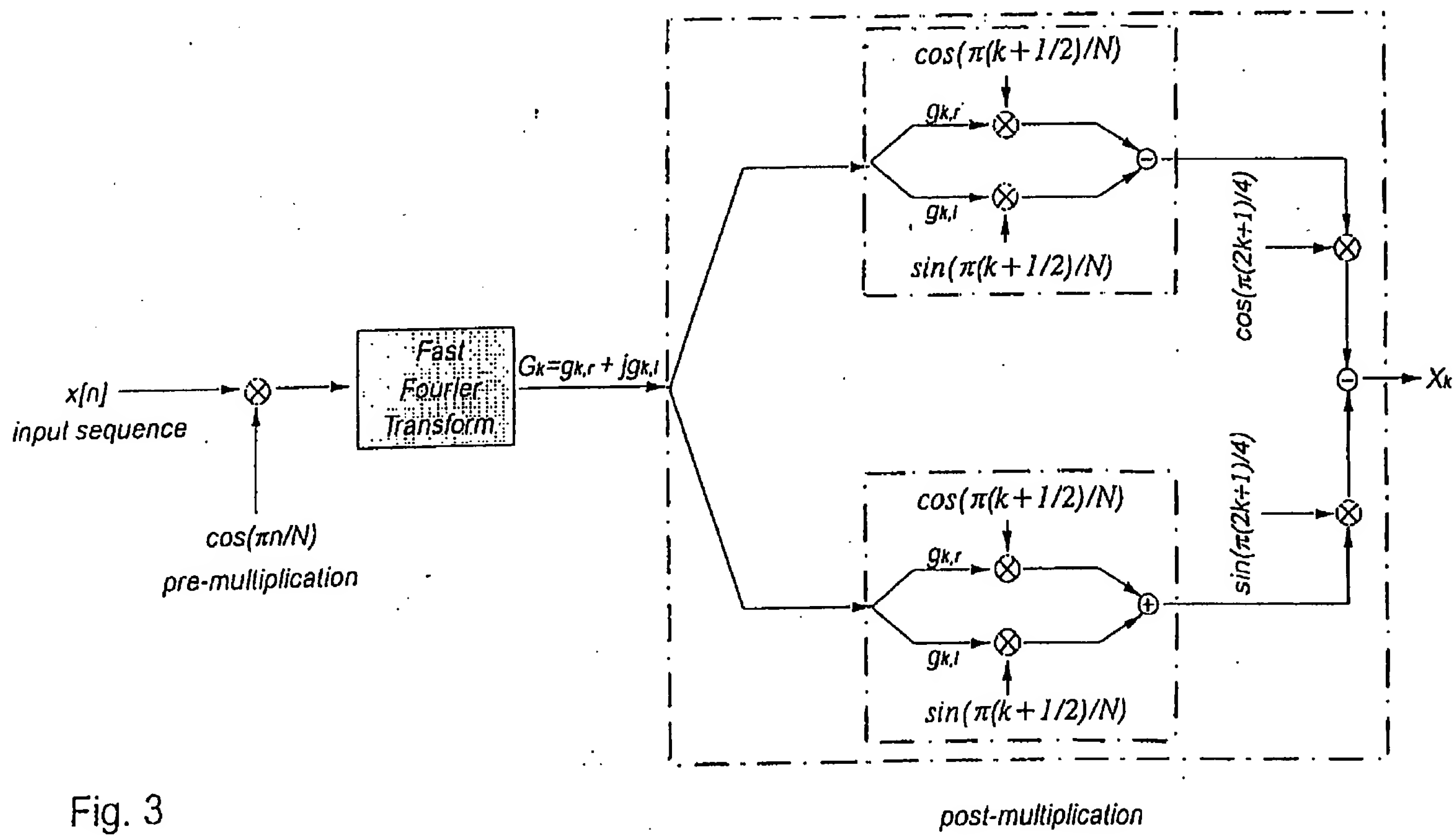

Fig. 3

AMENDED SHEET

APPENDIX F

RELATED PROCEEDINGS APPENDIX

THERE ARE NO RELATED PROCEEDINGS

851663.413USPC/805503v1

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